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PATENT & TRADEMARK OFFICE

8006-0013-52 PROV

TITLE OF THE INVENTION

A VIRTUAL ENVIRONMENT SOFTWARE ~~VOICE/COMPANY/OFFICE NETWORK~~ TOOL KIT, COMMON OPERATING CONTROL METHOD AND COMPUTER PROGRAM PRODUCT

Mark-Up #6

CROSS REFERENCE TO RELATED APPLICATIONS

The present document c … d in ~~co-pending U.S.~~
~~utility application Seria~~ … ~~ney docket number~~
~~8006-0006-52,~~ the issu … ly 11, 2000,
(hereinafter referred to … s of ~~which are being~~
incorporated herein by

Spec. Mark-Up
paper #6

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Field of the Invention

The present in … applications servers and computer-based … ssaging, (voice mail, e-mail, fax, instant messaging, etc.), electronic … ~~nd~~ the storage of electronic documents in secured locations, video and voice conferencing, plug in applications modules as well as disaster protection in a Virtual Environment hereby referred to as an application. ~~telephony disaster avoidance in a Virtual Voice/Company/Office Network environment.~~ In addition, these applications provide services that ~~Virtual environments~~ are differentiated from other services that allow telephony, multi-media messaging and or electronic document sharing and storage in that they typically employ CALL PULL-BACK technology to accomplish telephony Call Processing in the Public Switch Telephone Network (PSTN) and in the use of an applications tool kit and common operating control comprised of preprogrammed software constructs hereby referred to as "Objects."

~~These Virtual environments~~ Applications are housed in geographically diverse, strategically located ~~a~~ hardened sites called Network Operations Control Centers (NOCCS) and provide disaster protection by answering calls and allowing callers to be redirected to the called party's last known location. This is accomplished through the use of Presence. The called party logs on to the network from a Session Initiation Protocol (SIP) compatible device. The network makes note of the address of the device from which the call came and redirects future incoming traffic to that device until the next log on occurs. This may be used in conjunction with a type of "call follow me" and messaging. The network is private and the digital portion is encrypted. Everything is hardened against manmade or natural disasters and applications and messaging are mirrored from NOCC to NOCC. This is ~~avoidance when performing secondary answering~~ coupled with the ability to process calls from any device to any other device (analog, IP or cell phone, personal computer or PDA) ~~to locations such as key employees'~~

~~homes~~ in the event of an emergency. Multi-media messaging and/or document sharing is [[~~also~~]] used to seamlessly network together a client's staff even when that staff is deployed in multiple remote locations. More particularly, the present invention is directed [[~~to~~]] toward application development tools, ~~**T3I OBJECTS™;**~~ Objects and the methods used to ~~, methods and documentation used to market, deploy,~~ create, manipulate and/or destroy these Virtual Environments~~Voice/Company/Office Networks. Multi-media messaging may be accessed from a computer, telephone or related device.~~ Telephone calls are ~~may be~~ processed across a private IP network layered on top of a self healing optical network. ~~the Internet, Ethernet, Frame, ATM, Sonet etc. utilizing Voice over Internet Protocol, (VoIP).~~ Callers may be offered options when a call they initiate encounters a ~~*no answer*~~ no answer condition. ~~Virtual Voice/Company/Office Network applications operate on the hardware and software components comprising a Virtual Voice/Company/Office Network Node or Hub.~~

DISCUSSION OF THE BACKGROUND

Advances in communications (e.g., cellular telephones, PDAs and the Internet), the increased mobilization of the work force, the threat of manmade or natural disasters, and the desire of individuals to work securely from Virtual Environments ~~Offices~~ have all fueled the need for integrated communications services. These services often include the voice and data networking of employees and others working outside the traditional office environment. Subscribers may place calls, send, receive and manipulate multi-media messaging, share documents, and allow callers to access members of these networks no matter where they are located. Calls may be placed across the Public Switch Telephone Network (PSTN~~)~~) or a private IP network. ~~or the Internet, Ethernet, Frame, ATM, Sonet etc. utilizing VoIP.~~ The caller need never know that the person they are calling is working from a remote location ~~that may include their~~ or from home.

A client company or governmental unit's ("client") ~~customer's~~ ability to continue functioning after suffering a disaster is greatly enhanced since the equipment hosting their application[[s]] is located in multiple [[a]] hardened sites built to operate under extremely adverse conditions The digital portion of the network is layered on a self-healing optical network . Wherever possible, equipment is engineered to fail over to backup equipment. Applications and messaging are mirrored from NOCC to NOCC. The routing of callers to the main greeting of [[the]] a client's application ~~is handled~~ occurs at the ~~local~~ TELCO Central Office on a ~~*busy*~~ busy, forward or ~~*no answer*~~ no answer condition. If the original call was to a company rather than an individual, the caller could enter ~~The caller enters~~ an extension number or select[[s]] from a menu and [[is]] be transparently connected to an employee of the client. ~~subscriber~~. The caller is given further options if the call encounters a ~~*busy*~~ busy or ~~*no answer*~~ no answer condition. Clients ~~Subscribers~~ may access other clients ~~subscribers~~ on the corporate or government ~~their~~ network much like they ~~using a telephone or personal computer much like they~~ would in a traditional office._ The challenge has been to create the tools[[,]] and methods ~~and documentation~~ that enable the construction, maintenance and destruction of these networks in a rapid and reliable manner.

SUMMARY OF THE INVENTION

An objective of the present invention is to address the matters described in the Discussion of the Background. While the next section addresses specific features and attributes of the invention, a brief non-exhaustive description of the invention is now presented. The present invention provides a software tool kit; a computer based business method and services sold as products, which may be deployed by various means disclosed within this document.

~~While various system architectures are presented herein, one~~ The chief attribute of the invention is a software tool kit and common operating control known as ~~"T3I OBJECTS™; "~~ the Objects. In a previous version of the network, in the issued U.S. patent, Serial No. 6,088,437, dated July 11, 2000, the "Objects" resided in the Call Processor and Integrated Voice Response (IVR) portion of the NOCC. Today the "Objects" reside primarily in the SIP and Authentication servers. As the network evolves and technology changes, the "Objects" are rewritten. What remains the same or slowly changes over time is the underlying functionality of each Object. In general, the devices or servers that can be programmed and have the ability to issue commands to other devices or servers may be used.

Depending on the business needs of [[a]] the client, ~~OBJECTS~~ the Objects may work in conjunction with the invention described and disclosed in the issued U.S. patent, Serial No. 6,088,437, dated July 11, 2000 ("CALL PULL-BACK"), a copy of which is included in the Appendix. ~~co-pending U.S. utility application Serial No. 09/266,724 filed 03/12/99 bearing attorney docket number 8006-0006-52. OBJECTS~~ As previously mentioned, Objects are preprogrammed software constructs that when used in conjunction with one another allow a ~~serve as multi-use building block-like templates. These templates allow~~ non-technical person[[nel]] who may understand only the business needs of a client ~~customer~~ to rapidly and reliably create, manipulate and/or ~~accurately construct, maintain,~~ destroy an application. ~~and document a Virtual Voice/Company/ Office Network.~~These applications/Virtual Environments work in different ways. In some cases they stand behind the client answering calls that are forwarded in under a busy or no answer condition. In other cases the client may not have a physical headquarters. Callers go directly to the application and the application then routes calls and provides messaging, video and voice conferencing services.

While more common methods are used to market these products/applications, another attribute of the invention is to package the products as a boxed software package commonly found on store shelves and marketed on the Internet. The consumer purchasing the appropriate level of a Virtual Environment ~~Voice/Company/Office Network~~ from a store or Web Site may then order the desired configuration or upgrade needed on a Web site authorized to sell the products. The applications process switches circuit voice calls, H-320/ISDN video/voice conferencing traffic, IP/H-323 and Session Initiation Protocol (SIP) to and from a Media Gateway Control (MEGACO) cloud while the SIP presence is handled in the soft switch.

~~The products, Virtual Voice/Company/Office Networks, will process telephone calls, allow the user to access multi-media messaging from a phone or a computer, and allow document sharing which may be accessed through the web site of a client.~~

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete appreciation of the invention and many of the attendant advantages thereof will be readily apparent as the same becomes better understood. To assist in this understanding, the following is provided as a reference to the detailed descriptions to be considered in connection with the accompanying drawings located in the Appendix, wherein:

Figure 1 is a diagram of a method used to [[of]] implement[[ing]] the network. ~~a typical small Virtual Voice/Company/Office Network operating in a public switch telephone network (PSTN) environment;~~

Figure 2 is a diagram of one of the Network Operation Control Centers (NOCCS). ~~methods of implementing a typical small company Virtual Voice/Company/Office Network operating in various environments as shown in Figure 1 and Figure 2. While Figure 1 and Figure 2 are shown separately to simplify the concept of an average company, a Virtual Voice/Company/Office Network may employ both methods as well as using other services furnished by the Virtual Voice/Company/ Office Network Node or Hub. These services include all the services commonly employed by a National Internet Service Provider or National Internet Application Service Provider;~~

~~Figure 3 is a diagram of a method of implementing a Virtual Voice/Company/ Office Network Hub. A Hub is generally designed to handle the number of available customers in a geographic area covered by a carrier called a LATA. A given LATA could have more then one Hub or one Hub could cover more then one LATA. Hubs are connected to other Hubs by the Internet, Virtual Private Networks, Leased Lines, Wireless, Frame Relay, ATM, Sonet and other means;~~

~~Figure 4 is a detail diagram of a Node. (1) Switch. The number of Call Processing Servers on a particular switch, generally one per subnet, is dependent on the number of extension numbers on that switch that can be configured without the need for physical phones and associated equipment. (2) Subnets. (3) Routers. (4) Various servers;~~

~~Figure 5 is a diagram demonstrating how a caller reaches the correct Virtual application;~~

~~Figure 6 is a diagram of a method of utilizing a Sonet ring deployed in such a way as to allow callers in a LATA to reach the equipment centralized in one location where the desired application is hosted by dialing the minimum number of numbers required. In addition, this method makes it possible for inbound callers to reach the node-based application without incurring Interlata charges and for the processing of outbound calls from the node-based application within the LATA without incurring Interlata charges;~~

~~Figure 7 is a diagram showing the TELCO Central Office Nodes and Hub site placed on a counter rotating, self-healing Sonet ring that has a guaranteed downtime less then one second per failure. The ring provides transport for both voice and data in the~~

~~LATA;~~

~~Figure 8 is a diagram demonstrating a method of deploying equipment in multiple locations and using the carriers outbound footprint to provide service in a LATA;~~

~~Figure 9 is a diagram clarifying the functioning of objects in the Object Tool Kit.~~

BRIEF DESCRIPTION OF THE APPENDIX

An appendix is attached hereto that contains Figures 1 and 2 as well as a copy of the issued U.S. patent, Serial No. 6,088,437, dated July 11, 2000 ("CALL PULL-BACK") ~~through 9.~~

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring now to the drawing[[s]], wherein ~~like reference~~ numerals ~~designate~~ indicate ~~identical or~~ corresponding parts ~~throughout the views;~~ of the network:

~~Figure 1 is a diagram of a typical small company Virtual Voice/Company/Office Network operating in a public switch telephone network (PSTN) environment: (1) Source Telephone or VoIP Gateway. (2) PSTN Transport. (3) A caller directly dials the telephone number of a Virtual Voice/Company/Office Network or is forwarded in to the system because all of a customer's lines are busy, no one answers, or someone at the company transfers the caller. (4) The Virtual Voice/Company/Office Network Node or Hub contains the equipment hosting the application. (5) A D.I.D., D.N.I.S., A.N.I. or other number is sent from the Network and a translation on the incoming identification number may or may not be performed in order to accommodate a desired numbering plan. (6) The appropriate greeting is played to the caller.~~

Figure 1 is a diagram of the network.

(1) Represents the client's premise. Lines, trunks, time division multiplexing (TDM) or digital transport can provide connectivity to the network. A caller may trigger the application by directly dialing the DID or DNIS number, or by using caller ID, a pin number, or voice recognition. In the event DID, DNIS or caller ID is used, a translation on the incoming identification number may or may not be performed in order to accommodate a desired numbering plan. In addition, a caller may be forwarded into the application because all of the client's lines are busy, no one answers, or someone at the company transfers the caller in. Once the identifying string is received, the correct application is started and Objects provide the needed resources. The appropriate greeting is then played to the caller. ~~(7)~~The caller may send, receive or manipulate a mixed media message. ~~(8)~~The caller dials an extension, spells out some letters of a name, or otherwise makes a selection that dials a remote telephone number. ~~(9)~~The caller is processed across the PSTN, Internet, Leased Lines, wireless etc. to a remote destination telephone where CALL PULL-BACK may come into play. CALL PULL-BACK is disclosed in ~~co-pending U.S. utility application Serial No. 09/266,724 filed 03/12/99 bearing attorney docket number 8006-0006-52.~~, the issued U.S. patent, Serial No. 6,088,437, dated July 11, 2000, the entire contents of which are incorporated herein by reference. A client may or may not have a PBX. If they do have a PBX, the PBX may be programmed so that all or part of their traffic may use the network. A client may have a digital connection to the Point of Presence (POP), such as an Ethernet connection. If they do, switched circuit traffic may then be converted to digital traffic by a device located on or about the client's premise. In the event a caller does not have a PBX, IP Centrex or IP phones may be used. Video conferencing transport may be ISDN or IP. ~~Users may place a call to an analog telephone, a soft phone such as a properly equipped multi-media personal computer, or to a device such as a digital phone;~~

(2) Represents leased PRI connectivity to the Public Switched Telephone Network (PSTN).

(3) Multiple strategically located Class 5 switches in each LATA allowing all inbound and outbound traffic to be considered local traffic by the Incumbent Local Exchange Carrier (ILEC).

(4) Leased PRI tandem trunking connecting each Class 5 CO to . . .

(5) . . . a Gateway located in the Local Exchange Carriers (LECS) Class 4 Central Office or Data Center. The gateways are Points of Presence (POPs).

(10) The connection to a client's premise may be Session Initiated Protocol (SIP), H-323 over Ethernet or ISDN, all of which are native to the gateway eliminating any need for protocol converters and mediators.

(6) Represents a scalable, burstable Media Gateway Control (MEGACO) VPN to the IP cloud starting off with a T3 connection and expanding throughput as needed.

(7) IP cloud (MEGACO).

(8) Network Operations Control Center (NOCC) number 1. A NOCC also functions as a POP in the LATA in which it is located. NOCCs are connected by a self-healing optical network capable of withstanding a nuclear incident and spread out over large geographical distances. Each NOCC mirrors another NOCC's messaging and data. In addition, each NOCC provides a hot standby fail-over should a NOCC be rendered inoperable by a natural or manmade disaster. NOCCs contain the various servers and software needed to host the applications. (See figure 2 located in the appendix.)

(9) Network Operations Control Center (NOCC) number 2 provides services to its share of clients as well as mirroring the messaging and data of NOCC 1 and providing a hot standby fail-over for NOCC 1. To clarify, only NOCC 1 and 2 are shown on the drawing, but in reality there are as many NOCCs as are needed.

~~Figure 2 is a diagram of a typical small company Virtual Voice/Company/Office Network operating in a digital or Internet environment. (1) Various methods employed to interact with a Virtual Voice/Company/Office Network. (2) Transport. (3) The Virtual Voice/Company/Office Network Node or Hub contains the equipment hosting the application. (4) Switches, Subnets, Routers and Servers such as Gateway, Voice, Integrated Voice Response, and E-mail etc. that comprise a Virtual Voice/Office Node or Hub. (5) Users may send, receive or manipulate mixed media messaging. (6) Users may place a call from a soft phone such as a properly equipped multi-media personal computer or from a device such as a digital phone to either an analog or digital phone;~~

Figure 2 is a diagram clarifying the NOCC components,

(1) Incoming traffic from the POPs to the gateway.

(2) Media Gateway Control (MEGACO) Gateway.

(3) Gatekeeper: Pin prefix, account name, phone number, IP address, etc.

(4) Directory server: Central repository for storing and managing identity profiles and access privileges.

(5) SIP server which hosts SIP related applications.

(6) Unified Messaging Servers: Mixed media messaging.

(7) E-mail servers.

(8) Authentication and rating server, which hosts Softswitch applications.

(9) Common element manager: The Java-based graphical user interface (GUI) for managing individual network devices.

(10) Video and voice conferencing bridges.

(11) IVR IP Centrex server provides IP Centrex services for clients who have no PBX.

~~Figure 3 is a diagram of a Virtual Voice/Company/Office Network Hub. (1) Local Exchange Carrier (LEC). (2) VoIP transport. (3) Diagram of a 10 Node Hub. The number of Nodes in a Hub will vary depending on the capabilities of the Switch and the number of customers in a given area. (4) Node 1 of 10;~~

~~Figure 4 is a detail diagram of a Node. Routing may be accomplished at the LEC Network Level when more than one Node is in the Hub. Packet switching may be accomplished at the LEC, Point of Presence, or the Hub. In this manner, routing to the correct node and server is assured. (2) Subnet. (3) Router. (4) Represents one of the various Servers on the Subnet;~~

~~Figure 5 is a diagram demonstrating how a caller reaches the correct Virtual application. (1) Transport. (2) Central Office Node methods of routing for various services. (3) Identifying digits. (4) Translation of incoming digits from a fractured LATA numbering plan to the extension numbering plan of the Switch. Extension numbers are forwarded with digital integration information to the hunt pilot containing the digital ports of the Voice Server. (5) Port cross connections. (6) The correct application is started and the caller comes under control of the **T3I OBJECTS™;**~~

~~Figure 6 is a diagram of a method of utilizing a Sonet ring deployed in such a way as to allow callers in a LATA to reach the equipment placed in a centralized location. (1) Hub site hosting the desired application. (2) TELCO Central Office (CO) Node that callers can dial or be forwarded via a local number, (the minimum number of numbers required) without incurring Interlata charges. Time Division Multiplexing (TDM) may be employed so that callers may be *muxed* in by their local CO Node or gain access to the ring via a Gateway (TDM in; VoIP out). The VoIP method allows a many fold increase in throughput, and with VoIP over ATM over Sonet, data and video ride virtually free in~~

~~the packets needed to transport voice. Note that these TELCO nodes (CO Nodes) are located in various positions within the LATA so that subscribers may access them as a local call and then be "back hauled" across the ring to the centralized Hub location. (3) Pass through TELCO central office connecting the various TELCO Nodes on the ring as well as the CO Node local to the Hub location. (4) CO Node local to the Hub. Callers may be muxed out to the Hub by the local TELCO CO Node or routed to a packet switch and Gateway Servers;~~

~~Figure 7 is a diagram showing the TELCO Central Office Nodes and Hub site placed on a counter rotating, self healing Sonet ring. (1) Portrayal of the inner ring with traffic flowing in a clockwise direction. (2) Portrayal of the outer ring with traffic flowing in a counterclockwise direction. The Ring may provide a guaranteed downtime of less than one second per failure. (3) TELCO Node which users can reach as a local call and be "back hauled" to the Hub. (4) Hardened centralized equipment location called the Hub, which also contains the equipment for electronic document storage. To avoid a single point of failure the ring enters the Hub site in 2 places located on different faces of the Hub facility;~~

~~Figure 8 is a diagram demonstrating a multi-location four Node LATA method of providing service in a LATA. (1) Represents a LATA. (2) Nodes deployed at various locations in the LATA allowing all callers to access a Node in their area as a local call. The number of Nodes providing coverage in a LATA will vary with the demographics of the area. At this time in the competition for Interlata business, carriers are offering large toll-free outbound footprints. Inbound calls to the average Virtual environment customer are primarily all-local calls. Arrows from the Nodes (item 2) depict the outbound footprint. In most cases the outbound footprint is large enough to cover the majority if not all of the Nodes in a given LATA. These large outbound footprints allow the Nodes to process traffic without incurring Interlata charges. (3) PSTN, VoIP, Leased Lines, wireless, Fast Ethernet, Frame Relay, ATM, and Sonet etc. are some of the other means of transport that can be used to carry traffic between Nodes for callers who need to exceed the outbound footprint;~~

~~Figure 9 is a diagram clarifying the functioning of **OBJECTS** in the Object Tool Kit. (1) Incoming call from the PSTN. (2) Incoming VoIP call. For the sake of clarity, only one Subnet is shown in this diagram. (3) Switch that performs the numbering plan translations and is digitally integrated with the Call Processing Server (item 7), commonly referred to as the Voice Server. (4) Packet Controller/Switch and Gateway Servers etc. allowing VoIP callers access to the Switch (item 3) and routing data traffic to the Subnet. (5) Represents the Subnet. (6) Router. (7) Call Processing/Voice Server where callers come under control of **T3I OBJECTS™** which may in turn call upon services from the other Servers numbered 8-15 and the Router (item 6).~~

DESCRIPTION OF OBJECTS

An Object is a proven, preprogrammed software construct which by itself or when assembled with other Objects provides a desired functionality. ~~T3I OBJECTS™;~~ Objects are written in traditional programming languages, scripting languages, and high-level command line code. ~~T3I OBJECTS™~~ The Objects allow non-technical personnel who understand the business needs of a ~~customer~~ client to rapidly and accurately create, manipulate and/or destroy these ~~virtual environments~~ Virtual Environments.

~~T3I OBJECTS™~~ Objects operate in conjunction with parameters, tables, attributes, classes, routines, methods, compiled code etc. which control the various components in the [[Hubs]] NOCCs and [[Nodes]] POPs. ~~T3I OBJECTS™~~ Objects perform the various functions so that the needs of the client are [[met]] satisfied. The creator of a Virtual ~~environment~~ Environment places clients' mailboxes in Classes of Service that have been pre-configured as Objects. When services from other Servers are needed, ~~OBJECTS~~ the Objects furnish those services in the appropriate manner. If switching services are required, ~~OBJECTS~~ the Objects will issue commands to the Switch for functions such as routing a call. As with any product in the computer/telephony world, the ~~T3I Object Tool Kit is~~ Objects are constantly evolving.

Hardware changes require the rewriting of the "Objects" in new languages. Therefore, a ~~A~~ living document has been created describing the functionality and services provided by each Object rather than lines of code. Once the functionality is specified, it is a simple matter to rewrite each Object. ~~This documentation is used by personnel responsible for the creation of Virtual Environments.~~ Personnel responsible for the creation of Virtual Environment applications use the documentation that starts on page 14. The Objects used to create a given Virtual ~~environment~~ Environment have the explanation of the functionality and services of the Object under the heading of each Class of Service (COS) or Object (OBJ). The Object itself is not a Class of Service; it is all of the preprogrammed and tested software comprising that construct. This ability alone allows the offering of inexpensive, reliable, custom Virtual ~~environments~~ Environments in a very rapid and cost effective manner.

Cost effectiveness is an important reason that others are not building sophisticated custom configurations for large numbers of ~~small~~ clients. Without the use of ~~an Object Tool Kit~~ the Objects, their personnel would have to program each configuration from scratch. The best way to define a given Object is to define its functions. One of the unique features of ~~a T3i OBJECT~~ an Object is that a single instance of a given Object can operate at the same time on one or more physical platforms having different operating systems. Each Object is made up of many components.

~~Object/Class of Service documentation:~~ **Object/Class of Service documentation:**

OBJECT/ CLASS OF SERVICE	FUNCTION OF OBJECT/CLASS OF SERVICE
OBJ/COS 0	Unassigned D.I.D. mailboxes. NOTE: A silent D.I.D. mailbox greeting must be recorded. Non area code specific Object. Description: The number of a mailbox placed in this Object matches the number that will be received by the ~~Call Processor Portion~~ relevant portion of the [[Node]] POP or [[Hub]] NOCC after any and all translations are accomplished. This number is generated when a caller reaches a telephone number issued to a ~~customer~~ client. As numbers in Object 0 are not currently issued but are defined and can still be reached from the outside world, they are kept in Object 0. When one of these numbers is dialed, the following recording is played: "*You have reached an unassigned telephone number, please hang up and try again*". No input is accepted from the caller. No message is taken which prevents unwanted messages such as those left by automatic dialers from consuming storage. The caller hears, "Goodbye" and is dropped.
OBJ/COS 1	Company greetings, no associated extension or telephone number, no messages may be recorded. Play greeting twice and disconnect. Non area code specific Objects. Description: The caller hears a recorded message and may enter an extension number or select a menu choice. No numbers are dialed automatically upon a caller reaching this mailbox. If the caller takes no action, the greeting is played twice and the caller is dropped.
OBJ/COS 2	Disconnect from caller. Play silent greeting and hang up. No input is accepted from the caller. Non area code specific Object. Description: Commonly used as part of a configuration where there is a need to play a recording and while the recording is being played, allow the user to take an action such as entering an extension number or making a choice. If no selection or choice is made after the recording is played, there is a moment of silence while the caller is moved to a mailbox and placed in Object 2 where they may or may not hear a second recording depending on the application desired. No input is accepted from the caller. After the moment of silence or the second recording is played, the caller is dropped.

OBJ/COS 3	Company greetings, no associated extension or telephone number, no messages may be recorded. Play greeting and disconnect. Non area code specific Object. Description: The caller hears a recorded message and may enter an extension number or select a menu choice. If the caller takes no action, the caller is dropped.
OBJ/COS 4	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: If caller presses a 0, a *ring all* hunt group is activated.
OBJ/COS 5	Voice Response Server prompt mailboxes. Store recordings used by the Voice Response Server. Description: Store recordings used by the Voice Response (IVR) portion of the [[Node]] POP or [[Hub]] NOCC.
OBJ/COS 6	Pilot mailbox for Voice Response, (IVR) Server message pool. Store recordings used by the Voice Response Server application processor. Description: The lead mailbox number of a list of mailboxes responsible for the storage of application controlled messages.
OBJ/COS 7	Fax only with voice annotation. Non area code specific Object. Description: Any mailbox placed in this Object will accept only a fax with or without voice annotation. User input is accepted.
OBJ/COS 8	Voice Response Server Error mailbox. Store error recording used by the Voice Response Server, (IVR) application processor. Description: Any mailbox placed in this Object is an error-handling mailbox for the Voice Response Server, (IVR) portion of the [[Node]] POP or [[Hub]] NOCC. User input is accepted.
OBJ/COS 9	Fax on demand. Prints the first fax in each mailbox. Non area code specific Object. Description: Prints only the first fax stored in a mailbox placed in this Object.

OBJ/COS 10	Application Processor control. Non area code specific Object. Description: Takes the caller to Application Processor Control. A caller that reaches a mailbox in this Object is provided services by the Voice Response Server, (IVR) application associated with that mailbox which acts as a call identification number.
OBJ/COS 11	Plays greeting once; after greeting plays, use extension number for next mailbox. Non area code specific Object. Description: Normally used to play a recording once which may give the caller enough time to take an action such as dialing an extension or selecting a choice. After greeting plays or if the caller takes no action, the caller is moved to a different part of the application. Also used as a way to rapidly and automatically move a caller from one mailbox to another.
OBJ/COS 12	Play greeting twice; after greeting plays, use extension number for next mailbox. Non area code specific Object. Description: Normally used to play a recording twice, which may give the caller enough time to take an action such as dialing an extension or selecting a choice. After greeting plays or if the caller takes no action, the caller is moved to a different part of the application. Also used as a way to rapidly and automatically move a caller from one mailbox to another.
OBJ/COS 13	Block access to system distribution pilot numbers. Non area code specific Object. Description: Access to a distribution list whose pilot number is placed in this Object is restricted to users with special mailbox programming.
OBJ/COS 14	Fax overflow mailboxes (M/Bs). Non area code specific Object. Description: Used to provide a "fax store and forward" service to a ~~customer~~ client. When a fax machine on the ~~customer's~~ client's premise is ~~*busy*~~ busy or ~~*no answer*~~ no answer, the caller is forwarded to a mailbox in this Object that provides fax tone, takes a fax, and repeatedly attempts to deliver that fax back to the originally called fax machine until successful.

OBJ/COS 15	Call number first before playing greeting, record message option. Offsite only if urgent. May receive fax. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Ring an extension; if ~~*busy*~~ busy or ~~*no-answer*~~ no answer, play a greeting, take a message, and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 16	Call number first before playing greeting, record message option, station has multiple mailboxes; ask before connecting. Offsite only if urgent. May receive fax. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Ring a phone; if answered announce the call, if ~~*busy*~~ busy or ~~*no-answer*~~ no answer, play a greeting, take a message, and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 17	"Greeting on" stops numbers from being dialed, record message option. Offsite only if urgent. May receive fax. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Recording a greeting and turning the greeting on stops the extension number associated with a mailbox in this Object (if one exists) from being dialed. Mailbox will take a message and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 18	Professional voice. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Along with the normal features and functions given most users, a Mailbox placed in this Object has the privilege of being able to name mailboxes. Recording a greeting and turning the greeting on stops the extension number associated with a mailbox in this Object (if one exists) from being dialed. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 19	Forms. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: This Object provides specific ~~customers~~ customers with an application that asks a series of questions one at a time and records the answers the caller gives in their own voice. After the questions are asked, the caller is given the option of reviewing their answers and re-recording them if so desired. Upon acceptance of the answers by the caller, the answers to the questions are placed in a mailbox specified by the ~~customer~~ client for further action.

OBJ/COS 21	Page every time a message is left. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: When placed in this Object, a mailbox with an internal extension will activate a pager every time a message is left during user specified time periods. Recording a greeting and turning the greeting on stops the extension number associated with a mailbox in this Object (if one exists) from being dialed. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 30	Local call, call number first before playing greeting, record message option. Non area code specific Objects. Description: A mailbox placed in this Object will call an external telephone number without dialing an area code and if not answered, will play a greeting and record a message. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 31	Local call, call number first before playing greeting, record message option. Station has multiple mailboxes; ask before connecting. Non area code specific Objects. Description: Dial a telephone number without dialing an area code; if answered, announce the call; if ~~*busy*~~ busy or ~~*no-answer*~~ no answer, play a greeting and take a message. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 32	Local call, "greeting on" stops numbers from being dialed, record message option. Non area code specific Objects. Description: Recording a greeting and turning the greeting on stops the telephone number associated with a mailbox in this Object (if one exists) from being dialed. When the user turns off the greeting, a mailbox placed in this Object will ring a phone without dialing an area code. The mailbox will take a message. While listening to the greeting the caller may enter an extension number or select a choice.

OBJ/COS 33	Local call, call number first before playing greeting, record message option. Offsite only if urgent. May receive Fax. Non area code specific Objects. Description: Ring a phone without dialing an area code; if ~~busy~~ busy or ~~no answer,~~ no answer, play a greeting and take a message. Mailbox will take a message and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 34	Local call, call number first before playing greeting, record message option. Station has multiple mailboxes; ask before connecting. Offsite only if urgent. May receive Fax. Non area code specific Objects. Description: Ring a phone without dialing an area code. If answered, the call will be announced and the called party will be given the option to accept or reject the call. If ~~busy~~ busy or ~~no answer~~ no answer, a greeting may be played and a message taken. Mailbox will take a message and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 35	Local call, "greeting on" stops numbers from being dialed, record message option. Offsite only if urgent. May receive Fax. Non area code specific Objects. Description: Recording a greeting and turning the greeting on stops the telephone number associated with a mailbox in this Object (if one exists) from being dialed. When the user turns off the greeting, a mailbox placed in this Object will ring a phone without dialing an area code. Mailbox will take a message and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 36	Local call, blind transfer. Non area code specific Objects. Description: A mailbox in this Object will dial a telephone number without dialing an area code and then perform a blind transfer. The caller may dial no numbers and no messages may be recorded.

OBJ/COS 37	Local call, call number first before playing greeting, no messages, play greeting twice, allow user to dial. Non area code specific Objects. Description: A mailbox in this Object will dial a telephone number without dialing an area code; if ~~busy~~ busy or ~~no answer~~ no answer, the greeting will play twice, the user will be allowed to dial an extension or select a choice, no messages may be recorded.

The following is an example of a set of area code specific Objects. For clarity, only one set of area code specific Objects are shown.

OBJ/COS 50	(305) area code call; call number first before playing greeting, record message option. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. If the call is not answered the mailbox will play a greeting and record a message. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 51	(305) area code call; call number first before playing greeting, record message option. Station has multiple mailboxes; ask before connecting. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. If answered, it will announce the call; if ~~busy~~ busy or ~~no answer~~ no answer, it will play a greeting and take a message. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 52	(305) area code call; "greeting on" stops numbers from being dialed, record message option. Area code specific Objects. Description: Recording a greeting and turning the greeting on stops the telephone number associated with a mailbox in this Object (if one exists) from being dialed. When the greeting is turned off by the user, a mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. The mailbox will take a message. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 53	(305) area code call; call number first before playing greeting, record message option. Offsite only if urgent. May receive Fax. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. If ~~busy~~ busy or ~~no answer~~ no answer, play a greeting and take a message. Mailbox will take a message and activate "off site message waiting notification" if the message was marked urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.

OBJ/COS 54	(305) area code call; call number first before playing greeting, record message option. Station has multiple mailboxes; ask before connecting. Offsite only if urgent. May receive Fax. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. If answered, the call will be announced and the called party will be given the option to accept or reject the call. If ~~busy~~ busy or ~~no answer~~ no answer, a greeting may be played and a message taken. Mailbox will take a message and activate "off site message waiting if the message was marked urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 55	(305) area code call; "greeting on" stops numbers from being dialed, record message option. Offsite only if urgent. May receive Fax. Area code specific Objects. Description: Recording a greeting and turning the greeting on stops the telephone number associated with a mailbox in this Object (if one exists) from being dialed. When the greeting is turned off by the user, a mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. Mailbox will take a message and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 56	(305) area code call; blind transfer. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number and performing a blind transfer. No numbers may be dialed by the caller and no messages may be recorded.
OBJ/COS 57	(305) area code call; call number first before playing greeting, no messages, play greeting twice, allow user to dial. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. If the number is ~~busy~~ busy or ~~no answer~~ no answer, a greeting will play twice and the user will be allowed to dial an extension or select a choice; no messages may be recorded.

Additional ~~examples of~~ Objects:

OBJ/COS 455	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Ring a phone without dialing an area code. If answered, the call will be announced and the called party will be given the option to accept or reject the call. If ~~busy~~ busy or ~~no answer~~ no answer, a greeting may be played and a message taken. Mailbox will take a message and activate "off site message waiting" if the caller marked the message urgent. In addition, a mailbox placed in this Object will accept a fax. While listening to the greeting the caller may enter an extension number or select a choice. This Object provides a special dynamic call blocking service to the company. When a caller enters a universal port they are given a numeric value that stays with them for the duration of the call. The caller may only reach an Object with the same numeric value or a "0" value.
OBJ/COS 456	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Takes the caller to Application Processor Control. A caller that reaches a mailbox in this Object is taken to the IVR application associated with that mailbox which acts as a call identification number. This Object provides a special dynamic call blocking service to the company. When a caller enters a universal port they are given a numeric value that stays with them for the duration of the call. The caller may only reach an Object with the same numeric value or a "0" value.
OBJ/COS 457	(800) area code. Object dedicated to one of a kind, ~~customer~~ client specific application. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number. If answered, it will announce the call; if ~~busy~~ busy or ~~no answer~~ no answer, it will play a greeting and take a message. While listening to the greeting the caller may enter an extension number or select a choice. This Object provides a special dynamic call blocking service to the company. When a caller enters a universal port they are given a numeric value that stays with them for the duration of the call. The caller may only reach an Object with the same numeric value or a "0" value.

OBJ/COS 458	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Recording a greeting and turning the greeting on stops the telephone number associated with a mailbox in this Object (if one exists) from being dialed. When the user turns off the greeting, a mailbox placed in this Object will ring a phone without dialing an area code. Mailbox will take a message. While listening to the greeting the caller may enter an extension number or select a choice. This Object provides a special dynamic call blocking service to the company. When a caller enters a universal port they are given a numeric value that stays with them for the duration of the call. The caller may only reach an Object with the same numeric value or a "0" value.
OBJ/COS 459	Object dedicated to one of a kind, ~~customer~~ client specific application. Used to automatically move callers to different parts of an application. Description: This Object provides a special dynamic call blocking service to the company. When a caller enters a universal port they are given a numeric value that stays with them for the duration of the call. The caller may only reach an Object with the same numeric value or a "0" value.
OBJ/COS 460	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Plays an announcement twice and hangs up. This Object provides a special dynamic call blocking service to the company. When a caller enters a universal port they are given a numeric value that stays with them for the duration of the call. The caller may only reach an Object with the same numeric value or a "0" value.
OBJ/COS 462	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Ring a phone without dialing an area code; if answered, announce the call; if ~~*busy*~~ busy or ~~*no answer*~~ no answer, play a greeting and take the caller to a specific location in the ~~customer's~~ client's configuration. While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 463	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: Ring a phone without dialing an area code; if answered, announce the call; if ~~*busy*~~ busy or ~~*no answer*~~ no answer, play a greeting and take the caller to a specific location in the ~~customer's~~ client's configuration. While listening to the greeting the caller may enter an extension number or select a choice.

OBJ/COS 464	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: This Object provides a specific ~~customer~~ client with an application that asks a series of questions one at a time and records the answers that a caller gives in their own voice. After the questions are asked, the caller is given the option of reviewing their answers and re-recording them if so desired. Upon acceptance of the answers by the caller, the answers to the questions are placed in a mailbox specified by the ~~customer~~ client for further action.
OBJ/COS 465	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: This Object provides a specific ~~customer~~ client with an application that asks a series of questions one at a time and records the answers that a caller gives in their own voice. After the questions are asked, the caller is given the option of reviewing their answers and re-recording them if so desired. Upon acceptance of the answers by the caller, the answers to the questions are placed in a mailbox specified by the ~~customer~~ client for further action.
OBJ/COS 468	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: This Object provides time of day control so callers hear different appropriate recordings at different times of day.
OBJ/COS 469	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: This Object provides time of day control so callers who press a "0" for the operator will reach different mailboxes during different times of day and after hours. One mailbox will ring a phone without dialing an area code; if answered, announce the call; if ~~*busy*~~ busy or ~~*no answer*~~ no answer, play a greeting and take a message. The other mailbox takes a message without dialing a phone (normally used after hours or during lunch). While listening to the greeting the caller may enter an extension number or select a choice.
OBJ/COS 472	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: A mailbox placed in this Object has an intercept that can be controlled by time of day.
OBJ/COS 473	(718) area code; blind transfer. Area code specific Objects. Description: A mailbox placed in this Object will dial a "1" and the above area code before calling an external telephone number and performing a blind transfer. The caller may dial no numbers and no messages may be recorded.

OBJ/COS 474	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: This Object provides time of day control so callers hear different appropriate recordings at different times of day.
OBJ/COS 475	Object dedicated to one of a kind, ~~customer~~ client specific application. Description: The caller hears a recorded message and may enter an extension number or select a menu choice. No telephone numbers are dialed automatically upon a caller reaching this mailbox. If the caller takes no action, the greeting is played twice and the caller is dropped. A mailbox placed in this Object has a specific dedicated operator when a "0" is pressed.
OBJ/COS 511	Ports level Ports level Object. This is a fail safe Object where callers are sent when there is no identifying number coming from the PBX portion of the [[Node]] POP or [[Hub]] NOCC. This Object also comes into play when the Call Processor portion of the [[Node]] POP or [[Hub]] NOCC doesn't know what else to do with the caller due to a software or ring cadence error. This Object contains the greetings that are played during different times of day and the operator's mailboxes that are used when a caller presses "0".

The following is an example of ~~the records kept of~~ an actual ~~customer's~~ client's configuration with the names, addresses and telephone numbers changed to ~~insure~~ ensure the ~~customer's~~ client's privacy. Actual verbiage spoken to the caller with a three-letter name in place of the actual ~~customer's~~ client's name is also part of this example. Each client has a drawing of their configuration designed to be easily read by personnel responsible for the creation, manipulation and/or destruction of Virtual ~~Voice/Company/Office Networks~~ Environment applications. This also ~~Also~~ allows the drawing to be created rapidly and accurately by removing Objects from a master template drawing.

(See Drawing On Next Page)

Object-Use Example

XYZ Company
Contact-Name of person
Telephone (XXX) XXX-XXXX
Fax (XXX) XXX-XXXX
Street address
Suite X
City, State, Zip

1→(XXX) XXX-XXXX
2→M/B 2004,N,468 silent greeting
3→-MTWTF- 08:00-12:00 N/B 83500,N,1
-MTWTF-12:01-12:59 N/B 83600,N,1
-MTWTF-13:00-17:00 N/B 83500,N,1
Default N/B 83700,N,1
4→Business hours Intercept M/B 83800,9979390,32
5→A/C A/E to M/B 1077,9979390,N,32
6→Lunch and Default Intercept M/B 83900,N,32
A/C A/E to M/B 1077,9979390,32

M/B 83500,N,469
Intercept M/B 83800,9979390,32
7→A/C A/E to M/B 1077,9979390,32
Max 1 Pre 8410

M/B 83600,N,469
Intercept M/B 83900,N,32
A/C A/E to M/B 1077,9979390,32
Max 1 Pre 8410

M/B 83700,N,469
Intercept M/B 83900,N,32
A/C A/E to M/B 1077,9979390,32
Max 1 Pre 8410

M/B 84101,N,469
Business hours intercept M/B 83800, 9979390,32
A/C A/E to M/B 1077,9979390,32
Lunch and Default intercept M/B 83900,N,32
A/C A/E to M/B 1077,9979390,32

Application located in NOCC 1
Default security code KUNIK or 58645
Voice Doreen Panico
User transfer on all trunks. Call forward on R/N/A on all trunks

1. (XXX) XXX-XXXX is the telephone number that receives callers who are forwarded or transferred in from a ~~customer~~ client location or who dial in directly.

2. M/B 2004,N,468 is a mailbox whose first four numbers match the last four digits of the telephone number that receives callers. 2004 is the mailbox number. N means there is no telephone number or extension number associated with this mailbox as a number to be dialed when the caller reaches mailbox 2004. 468 is the Object number. M/B 2004 has been placed in Object 468. Object 468 is dedicated to a one of a kind, ~~customer~~ client specific application. Object 468 provides time of day control so callers hear different appropriate recordings at different times of day.

3. Monday through Friday, (-MTWTF-) from 08:00 A.M - 12:00 P.M. a moment of silence is played to the caller. The caller then hears the business hours greeting stored in mailbox 83500,N,1. Upon hearing any part of the greeting recorded in mailbox 83500, the caller may enter an extension number. This extension number is in reality a mailbox number. The caller may also select a choice that may be offered in the recording played to the caller such as 0 or 1-9. If the caller does nothing, the greeting will repeat and after several seconds of silence, the caller will hear "Goodbye" and be dropped. The caller will not be permitted to leave a message.

In the above table, Object 1 has the following definition:

Company greetings, no associated extension or telephone number, no messages may be recorded. Play greeting twice and disconnect. Non area code specific Object. Description: The caller hears a recorded message and may enter an extension number or select a menu choice. No numbers are dialed automatically upon a caller reaching this mailbox. If the caller takes no action, the greeting is played twice and the caller is dropped.

4. When a mailbox such as (M/B) 83800,9979390,32 is reached by a caller selecting choice "0" which is the business hours intercept (the operator), the Object obtains the appropriate external dial tone and dials the telephone number 997-9390. Once the telephone number is dialed, CALL PULL-BACK is employed. (See ~~Utility patent pending number 8006-0006-52 CALL PROCESSING SYSTEM, METHOD AND COMPUTER PROGRAM PRODUCT.)~~ issued U.S. Patent, Serial No. 6,088,437, dated July 11, 2000, incorporated herein by reference.)

5. A/C A/E to M/B 1077,9979390,32 means that if a caller leaves a message in M/B 83800, it will be "auto copied" to M/B 1077 and "auto erased" from M/B 83800. M/B 1077 will obtain the appropriate external dial tone from its Object, and depending on how the ~~customer~~ client wants the messaging waiting notification times set up in that M/B, message waiting notification will be performed to the telephone number 997-9390. At different times of day different ~~company~~ client main greetings are played to the caller.

6. Default means all other times not specified and is typically used for after hours.

7. Max 1 Pre 8410 means that if the caller presses choice "1", they will be taken to M/B 84101 and governed by Object 469.

The recordings spoken to the callers by a mailbox (M/B) number are as follows:

XYZ Company Greetings

M/B 83500

Business Hours: Monday through Friday 8:00 A.M. to 12:00 P.M.
1:00 P.M. to 5:00 P.M.

Greeting: *Thank you for calling XYZ Company. All available phone lines are utilized or our operator is assisting a previous caller. If you know your party's extension, please enter it now or press "0" for the operator. For our corporate directory, press "1".*

M/B 83500, 83600 & 83700

Holiday Greeting. Put in as a message and prior to the holiday, do a greeting/message swap.

Thank you for calling XYZ Company. Our offices are closed for the holiday. If you would like to leave a message, we will be checking in; however, the operator will not be available to assist you. For the corporate directory, press "1". Have a great holiday.

M/B 83600

Lunch Time: Monday - Friday 12:01 P.M. to 12:59 P.M.

Thank you for calling XYZ Company. We are closed for lunch and will return at 1:00 P.M. If you know the extension of the person for whom you wish to leave a message, please enter it now. For our corporate directory, press "1". To leave a message for our operator, press "0".

M/B 83700

After hours

Thank you for calling XYZ Company. Our office hours are Monday through Friday from 8:00 A.M. to 5:00 P.M. If you know the extension of the person for whom you wish to leave a message, please enter it now. For our corporate directory, press "1". To leave a message for our operator, press "0".

M/B 83800

Name *The receptionist*

M/B 83900

Name *The receptionist*

Greeting: *Please leave a message at the sound of the tone and we will return your call as soon as possible.*

M/B 84101

Greeting: *The following is a list of our corporate personnel. At any time you may enter their extension number to leave a message.*

First Name	Last Name	extension 1060
First Name	Last Name	extension 1061
First Name	Last Name	extension 1062
First Name	Last Name	extension 1063
First Name	Last Name	extension 1064
First Name	Last Name	extension 1065

First Name	Last Name	extension 1066
First Name	Last Name	extension 1067
First Name	Last Name	extension 1068
First Name	Last Name	extension 1069
First Name	Last Name	extension 1070
First Name	Last Name	extension 1071
First Name	Last Name	extension 1072
First Name	Last Name	extension 1073
First Name	Last Name	extension 1074
First Name	Last Name	extension 1075
First Name	Last Name	extension 1076
The receptionist		extension 1077

M/B 1077

Name *The receptionist*

Greeting: *You have reached (First Name, Last Name), the Receptionist. If you are calling concerning an office matter or to schedule an appointment, please leave your name and telephone number at the tone and we will get back to you.*

M/B 1060

Greeting: *You have reached the voice mail of (First Name, Last Name). If you are calling concerning an office matter or to schedule an appointment, please call (First Name) at extension 1077 by pressing "1" now. If this is urgent, you may leave a one-minute voice message that will page (First Name). Please leave your message after the tone.*

All M/Bs in the following list;

First Name	Last Name	M/B 1069
First Name	Last Name	M/B 1061
First Name	Last Name	M/B 1062
First Name	Last Name	M/B 1063
First Name	Last Name	M/B 1064
First Name	Last Name	M/B 1065
First Name	Last Name	M/B 1067

First Name	Last Name	M/B 1068
First Name	Last Name	M/B 1070
First Name	Last Name	M/B 1071
First Name	Last Name	M/B 1073
First Name	Last Name	M/B 1074
First Name	Last Name	M/B 1075
First Name	Last Name	M/B 1076

Greeting: *You have reached the voice mail of (First Name, Last Name). Please leave a detailed message at the sound of the tone and your call will be returned as soon as possible.*

All M/Bs in the following list;

First Name	Last Name	M/B 1066
First Name	Last Name	M/B 1072

Greeting: *You have reached the voice mail of (First Name, Last Name). Please leave a detailed message at the sound of the tone and your call will be returned as soon as possible. For further options, press star* after your message. If you mark your message urgent, (First Name, Last Name) will be paged.*

The processes set forth in the present description may be implemented using a conventional general purpose microprocessor programmed according to the teachings of the present specification, as will be appreciated to those skilled in the relevant art(s). Appropriate software coding can readily be prepared by skilled programmers based on the teachings of the present disclosure, as will also be apparent to those skilled in the relevant art(s).

The present invention thus also includes a computer-based product that may be hosted on a storage medium and may include instructions, which can be used to program a computer to perform a process in accordance with the present invention. The storage medium can include, but is not limited to, any type of disk including floppy disk, optical disk, CD-ROMS, and magneto-optical disks, ROMS, RAMs, EPROM's, EPROM's, flash memory, magnetic or optical cards, or any type of media suitable for storing electronic instructions.– Numerous modifications and variations of the present invention are possible in light of the above teachings and should be construed as part of the present invention.

~~Claims:~~ CLAIMS

1. (Currently Amended) A method of configuring a communications system utilizing CALL PULL-BACK technology as disclosed in ~~co-pending U.S. utility application Serial No. 09/266,724 filed 03/12/99 bearing attorney docket number 8006-0006-52,~~ the issued U.S. patent, Serial No. 6,088,437, dated July 11, 2000. The Objects are first disclosed in the ABSTRACT OF THE DISCLOSURE page 33, line 16, of the above referenced issued patent CALL PROCESSING, METHOD AND COMPUTER PROGRAM PRODUCT as preprogrammed and proven software constructs. Over time, hardware changes require the rewriting of the Objects in new languages. The functionality of the OBJECTS is defined in the **Object/Class of Service documentation, commencing on page 15 of this application,** incorporated herein by reference as though set forth in full. Once the functionality of each Object is known, it is a simple matter to rewrite each Object as needed. This method is comprised of ~~comprising~~ the following steps:

~~preparing~~ Preparing Objects as preprogrammed software constructs, said Objects being configured to perform predetermined functions when populated with a set of user definable parameters subsequently executed by processors;

~~inputting~~ Inputting said set of user definable parameters into said Objects so as to perform said predetermined functions when executed by said processors.

2. (Currently Amended) A reconfigurable communications system, ~~comprising: processors;~~ comprising processors data and voice input devices; memory encoded with Objects as preprogrammed software constructs, said Objects being configured to perform predetermined functions when populated with a set of user definable parameters input by said input devices and subsequently executed by the processors.

3. (Currently Amended) A business method of custom configuring a communication system, comprising the following steps:

~~collecting~~ Collecting a set of communication system attributes from a predetermined source;

~~preparing~~ Preparing an Object as a preprogrammed software construct, said Object being configured to perform a predetermined function when populated with said set of communication system attributes and subsequently executed by a processor;

~~inputting~~ Inputting said set of communication system attributes into said Object so as to perform said predetermined function when executed by said processor.

4. (Original) The business method of Claim 3, wherein: said step of collecting comprises collecting said set of communication system attributes from a ~~customer~~ client.

5. (Currently Amended) Methods of utilizing Objects by non-technical personnel, who understand the business needs of a client to create, manipulate and/or destroy Virtual ~~Voice/Company/Office Networks.~~ Environment applications.

6. (Currently Amended) A method of configuring a communications system utilizing CALL PULL-BACK and Objects technology to process PSTN and VoIP callers utilizing:

Analog, IP or cell phones, personal computer or PDAs ~~Digital, Computer, telephones or properly equipped personal computers, (Soft Phones)~~ with one device being able to talk to any other of these devices;

~~Packet or Soft Switches, Packet Controllers, VoIP Gateways and Routers; NISP, RISP, ISP or ASP Service Providers;~~

Gateways, Gatekeepers, Directory servers, SIP servers, Unified Messaging servers, E-

mail servers, Authentication and rating servers, Common Element Managers, Video and voice conferencing bridges and IVR IP Centrex servers;

~~leased~~ Leased lines [[or]] wireless~~; PPP, FDDI, Fast Ethernet, Frame Relay or Sonet~~ TDM; Ethernet or VPNS as transport.

7. (Currently Amended) A method of configuring a communications system:

~~utilizing~~ Utilizing CALL PULL-BACK and Objects technology to provide services such as[[:]] telephony, multi-media messaging, (voice mail, e-mail, fax, instant messaging etc.); electronic document-sharing, [[and]] the storage of electronic documents in secured locations, video and voice conferencing, plug in applications modules as well as disaster protection in a Virtual Environment application; ~~networking together a clients staff even when that staff is deployed in multiple remote locations.~~

8. (Currently Amended) A method of configuring a communications system utilizing CALL PULL-BACK and Objects technology to provide disaster avoidance by processing callers to key employees' homes or alternate sites, or who dial or are forwarded in to the system from the TELCO Central Office under various *no answer* conditions. ~~This method may be accompanied by the storing of electronic documents at a secure site.~~

9. (Currently Amended) A method of utilizing hardened [[Hub]] POP and NOCC sites with back up power, equipment redundancy, and multiple voice and data transport connections, a meshed Optical Network to configure a highly disaster-resistant communications system utilizing CALL PULL-BACK and Objects technology.

10. (Withdrawn) A method of utilizing Nodes within the Hub sites and utilizing TELCO level or Hub level routing to assure traffic is directed to the appropriate Switch, Server and Router in a communications system utilizing CALL PULL-BACK and Objects technology.

11. (Withdrawn) A method of utilizing Subnets containing the various Routers and Servers to furnish the desired services required in a communications system utilizing CALL PULL-BACK and Objects technology.

12. (Withdrawn) A method of assuring a match from a fractured LATA numbering plan to the Node switch numbering plan in a communications system utilizing CALL PULL-BACK and Objects technology.

13. (Withdrawn) A method of providing a digital integration between the Node or Hub Switch with the Call Processing/Voice server in a communications system utilizing CALL PULL-BACK and Objects technology.

~~14.~~ 10. (Currently Amended) (Was claim 14) A Marketing method consisting of packaging the Virtual~~/Voice/Company/Office Network~~ Environment applications products as computer software commonly found in boxes on store shelves and marketed on the Internet. The consumer purchasing the appropriate level of a Virtual ~~Voice/Company/Office Network~~ Environment applications from a store or Web Site may then order the desired configuration or upgrade needed on a Web site authorized to sell the products.

~~15.~~ 11. (Currently Amended) (Was claim 15) A method of configuring a communications system utilizing CALL PULL-BACK, Objects technology and a self healing optical network as transport for VOIP, Video Conferencing, messaging and data traffic allowing users to access a local class 5 switch without incurring Interlata or long distance charges. ~~Sonet Ring as transport for two way PSTN, VoIP, and Data traffic allowing user traffic to access a local CO Node without incurring Interlata or long distance charges.~~

~~16.~~ 12. (Currently Amended) (Was claim 16) A method of configuring a

communications system utilizing CALL PULL-BACK, Objects technology, and a self healing optical Network ~~Sonet Ring~~ to ensure a minimum amount of downtime ~~stated as less then one second per failure~~.

~~17.~~ 13. (Currently Amended) (Was claim 17) A method of configuring a communications system utilizing CALL PULL-BACK, Objects technology, and a Meshed Optical Network ~~Sonet Ring configured with IP over ATM over Sonet~~ as transport for ~~packetized traffic.~~ VOIP, Video Conferencing, messaging and data traffic.

~~18.~~ 14. (Currently Amended) (Was claim 18) A method of configuring a communications system utilizing CALL PULL-BACK, Objects technology, and [[Nodes]] POPs deployed at strategic Class 5 switches and Data Centers ~~various locations~~ in the LATA allowing ~~users Node access as a local call. Large outbound footprints are used to process outbound traffic. When required for callers who need to exceed the outbound footprint PSTN, VoIP, Leased Lines, wireless, Fast Ethernet, Frame Relay, ATM, and Sonet etc. are some of the other means of transport that can be use to carry traffic between Nodes.~~ all traffic to be considered local traffic.

~~19.~~ 15. (Currently Amended) (Was claim 19) A method of record keeping for each client that produces a drawing of their configuration designed to be easily read by personnel responsible for the creation, manipulation and/or destruction of Virtual ~~Voice/Company/Office Networks.~~ Environment applications. This method allows the drawing to be rapidly and accurately created by removing Objects from a master drawing.

~~20.~~ 16. (Currently Amended) (Was claim 20) A method of documenting the spoken verbiage and user of a Virtual~~/Voice/Company/Office Network.~~ Environment application.

~~21.~~ 17. (Currently Amended) (Was claim 21) A method of documenting each Object used by the personnel responsible for the creation, manipulation and/or destruction of Virtual ~~Voice/Company/Office Networks.~~ Environment application.

ABSTRACT OF THE DISCLOSURE

The use of software-based Objects allows the rapid and accurate configuration, manipulation and/or destruction of Virtual ~~Environments~~ Environment applications that networks subscribers together, processes calls, enables messaging, and provides disaster avoidance.

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TITLE OF THE INVENTION

CALL PROCESSING SYSTEM, METHOD AND COMPUTER PROGRAM PRODUCT

CROSS REFERENCE TO RELATED APPLICATIONS

The present document claims the benefit of the earlier filing date of, and contains subject matter related to that disclosed in, co-pending U.S. provisional application Serial No. 60/082,730 filed April 23, 1998, having common inventorship, the entire contents of which being incorporated herein by reference.

BACKGROUND OF THE INVENTION

Field of the Invention:

The present invention pertains to call processing systems, methods and computer-based products used for telephony systems in which calls may be screened by a called party prior to connection. More particularly, the present invention is directed to voice and data systems that include Call Processors and Gateway Servers among other data communication resources, in which calls targeted for a predetermined location are directed by the programming of a Virtual Voice Network. This direction can be to any device that can be directly dialed, such as a telephone including cell phone, fax machine or modem even if the call for the targeted device is processed across the Internet, (voice or fax over Internet Protocol.)

Discussion of the Background

Advances in modern electronics and digital communication enable individuals to communicate virtually anywhere around the world. With the advent of cellular telephones,

personal communication services, and satellite telephony, individuals in advanced as well as developing societies have an expectation of being able to communicate with others anytime and anywhere in a smooth and seamless fashion. The bulk of existing communication infrastructure is provided by local, regional and long distance telephone companies, i.e., the public switch telephone network (PSTN), which uses land lines, among other resources, for providing point-to-point communications, where each point is identifiable by a separate telephone number. For example, a caller may use a first telephone number when attempting to reach a person at the person's home, but uses a second number for contacting the person at the person's office.

A recent challenge has been how to use the PSTN, components of which contain old technology, to provide the flexibility to support people who want to remain accessible while being mobile. In light of this backdrop, some corporations use private branch exchanges (PBX) at the corporation facilities to provide "smart" functions for handling incoming phone calls (described generically here, but referring to both voice and data calls) for the convenience of its customers. Using these functions, even if an employee is not available, a properly equipped PBX enables an outside caller to be conveniently patched into voice mail, routed to another number, or perhaps transferred to a different facility in an attempt to handle the call in a user-friendly environment.

Figure 1 is a block diagram of a conventional PSTN and PBX based system that enables a source telephone 1 to communicate with a destination telephone 11, of an intended recipient. In Figure 1, the source telephone 1 connects to the PSTN 3 via a line (wired or wireless). The PSTN 3 recognizes the telephone number input at the source telephone 1 and provides the switch infrastructure to ultimately connect the caller with the PBX 9, which has the burden of providing the "operator" interface functions. In many cases, the corporation facilities 7 may incorporate a relay Call Processor with auto attendant functions 13 even though unit costs for such devices could exceed 1.5 million dollars in 1998. An example of such a Call Processor is an

OVERTURE 300 sold by the Lucent/Octel Messaging Division.

The relay Call Processor with auto attendant 13 operates when the call is received by the PBX 9 and attempts to ring the destination telephone 11, while placing the caller on hold or utilizing any of a number of types of integration depending on the makes and models of the equipment. If the destination telephone 11 is not picked-up after a predetermined number of rings, the Call Processor with auto attendant 13 initially reports a message to the caller such as "thank you for calling company A. John Doe is on the telephone so please leave a message, dial another extension, or dial 0 for the operator." The Call Processor with auto attendant 13 is able to handle the call for the employee in this way because the PBX 9 receives and routes all telephone calls within the company facility 7 without having to interface with a variety of different local telephone equipment, each having unique signaling attributes.

For users that do not have the benefit of a corporation's PBX 9, the PSTN offers users a call forwarding operator 5 that, at the instruction of the intended recipient, forwards incoming calls to a secondary number when the intended recipient is unavailable at a primary number. This call forwarding mechanism however employs equipment at the PSTN and does not offer the same degree of convenient voice mail and auto attendant functions offered by the PBX 9 at the company facility 7.

As presently recognized by the inventor, the PBX 9 is an inherently "local" device hosted at a certain destination facility, such as a company. Available for equipment of such expense, smaller devices such as the relay Call Processor with auto attendant 13 are included with the PBX 9 to provide added functionality. Adding to the expense, the relay Call Processor with auto attendant 13 must be customized by technicians when installed at the company facility 7 so as to be compatible with the local telephone company equipment if any screened transfer types of calls were to be placed to external telephone numbers. Customization is needed because the PSTN 3 is not homogenous, but rather made up of numerous equipment of local telephone companies that

may or may not have the same equipment. As an example of different signaling attributes of signals provided by typical telephone equipment, the frequency and cadence of slow-busy signals (or other signals, as will be discussed) may be substantially different from one local telephone to the next. Similarly, other signals such as a fast busy signal, indicating an error is present, differs as well.

Figure 2 is a timing diagram of a ring/silence signal offered by exemplary local telephone company equipment. A high voltage level indicates a ring interval, while a lower voltage indicates a silence interval. For illustrative purposes, the interval "A" may typically range between a maximum tone-on (i.e., ring interval) of 1,200 ms to a minimum of 800 ms, while a typical number may be 1000 ms. The interval "B"(a silence interval) may range between 3500 ms and 2801 ms, with a typical number being 2881 ms. Interval "C" may typically range between 1200 ms and 800 ms, with a typical number being 942 ms. Interval "D" may typically range between 3485 ms and 2899 ms, with a typical number being 2910 ms. Similarly, the interval "E" may typically range between 1200 and 800 ms with a typical time 785 ms (which is less than the stated lower end of the "typical" range, but included to show that it is nonetheless a possibility). Due to this variation in cadence and frequency between signals provided by local telephone equipment, generic relay Call Processors with auto attendant functions are conventionally believed to require the use of technicians to personally customize the "application delays". This approach essentially normalizes the cadence and frequency terms so that the Call Processor can effectively interface with that particular local telephone equipment. Consequently, according to conventional wisdom, it is not believed wise, nor even possible, to use a relay Call Processor with auto attendant function in a central location that operates with different local telephone equipment because the diversity of telephone equipment does not permit the relay Call Processor with auto attendant to handle common signals in a like fashion.

Figure 3 is a flowchart of an example method of how a caller at a source telephone 1 (Fig.

1) attempts to communicate with an intended recipient at the company facility 7. The process begins in step S1, where the caller initiates a call to the intended recipient by dialing a phone number of the company where the intended recipient is believed to be located. The process then proceeds to step S3, where the call is answered by the PBX 9 at the company facility 7, and the PBX 9 passes the call to the Call Processor with auto attendant 13. The process then proceeds to step S5, where the caller is requested to dial the extension of the intended recipient. The process then proceeds to step S7 where an inquiry is made regarding whether the individual identified at that extension is available. While making the inquiry, the Call Processor in the PBX 9 places the caller on hold and rings the destination telephone a predetermined number of times. If the intended recipient does not answer the telephone call after the predetermined number of times or if a busy signal is received, the Call Processor concludes that the intended recipient is unavailable. If the response to the inquiry in step S7 is affirmative, the PBX 9 connects the caller with the intended recipient in step S9 and the process then proceeds to step S11 where the call is completed and then the communication session ends. However, if the response to the inquiry in step S7 is negative, the process proceeds to step S13 where the relay Call Processor with auto attendant 13, audibly presents a set of options to the caller. Typical options include leaving a voice mail message, hitting zero to dial an operator or entering the extension of another party. Once the options are presented, the process proceeds to S15 where the caller selects an option and then in step S17 the selected option is executed. Subsequently the process ends. The Call Processor may also offer other options to the caller, such as attempting to contact the intended recipient at another location. If the caller chooses this option, the Call Processor with auto attendant 13 performs a blind transfer to that other location. Since the Call Processor with auto attendant 13 performs the blind transfer, the Call Processor performs no additional processing of the call even if the intended recipient is not available at the other location. The blind transfer will be made regardless; even if the called party is busy, ring no answer, error tone or dead air.

Some coverage methods employ various call forwarding schemes in the event a called device is busy, or ring no answer. These methods are designed to forward the caller to a receptor mailbox. Often the receptor mailbox is located where the call originated and the called party pays the bill for any forwarding or long distance charges.

As identified by the present inventor, a limitation with conventional devices and methods is that the functions offered by the Call Processor in the PBX 9 are prohibitively expensive for the "small user". In other words, the call processing functions available at the company facility 7 are expensive to purchase and install, and thus are unsuitable for private use. Furthermore, due to differences between the different types of local telephone company equipment employed throughout the PSTN, making a conventional relay Call Processor with auto attendant available to users across a number of different local telephone company equipment is not believed to be possible, due to the different signaling attributes of the equipment employed by the different local telephone companies.

As presently recognized, the installation procedures of PBX 9 with the relay Call Processor are complex in that "hands on" customization and testing of the local telephone equipment is believed to be required in conventional systems when adjusting the destination delays for the relay Call Processor. Such difficulties are factors that contribute to the expense of purchasing and maintaining a Call Processor, even though conventional Call Processors are used over a specific geographical region sharing a common set of telephone equipment.

More centralized functions, such as call forwarding operations provided by the PSTN are incapable of detecting whether a person is available at one of the candidate locations, and "pulling back" the call for further processing if the person is unavailable. Moreover, the call forwarding operations perform a blind transfer of the call, and do not wait to determine whether the destination party will in fact receive the call. As a consequence, the user-friendliness of the call forwarding operation is presently viewed as being sub-optimal.

U.S. Patent No. 5,375,161 describes a telephone control system with branch routing, which includes a call conferencing feature (see, e.g., Figure 14', step 1419) that waits to determine whether or not a user may be located at another number. This technique thus employs precision busy/ring detection that requires *a priori* knowledge of the attributes of the local telephone communication equipment. Without this knowledge, it would not be possible for such a device to operate without significant customization of PSTN equipment at varying locations. Furthermore, the precision busy/ring would not be able to recognize an error tone, which has the same frequency as a busy but a different cadence, because the precision busy/ring unit monitors only frequency, not cadence. Cadence is a variable that fluctuates most from Central Office to Central Office. When considering a conference feature with voice, a number of conditions should be taken into consideration, including hardware sensitivity and ability to be configured, susceptibility to talk off in which the human voice emulates touch tone, and background noise. Regarding background noise, a PC modem, for example, connecting to a service provider for Internet access can cause any touch-tone activated equipment to do unexpected things. Thus, conferencing features are suboptimal.

SUMMARY OF THE INVENTION

Accordingly, a feature of the present invention is to provide a novel system, method and computer based product that overcomes the limitations of the conventional methods and systems discussed above. While a full description of the invention and its various features are described in the following section, a brief, non-exhaustive description of features of the present invention is now described. A facet of this invention is that the Call Processors used normally do not reside at the same location and are not directly connected to a customer's PBX or to the customer's Central Office. If the called party is busy / no answer or an error tone is encountered, the calling party is informed of the status of the called party and may be offered further options.

A "CALL PULLBACK" mechanism is included in a central location (i.e., accessible to geographically separated users) in a Call Processor, which is a component of a virtual Call Processor network. The Call Processor in the virtual Call Processor network places a caller on "soft hold" while attempting to contact the intended recipient at one of various predetermined numbers. In order to overcome the incompatibility issue of operating with different local telephone equipment, a feature of the present invention is a frequency and cadence detection mechanism that is able to detect different characteristics of slow busy, fast busy, ringing, answered, and ring no answer tones as provided by different local telephone equipment. To this end, the Call Processor of the present invention associates different frequencies and cadences with various events occurring with candidate numbers at which the intended recipient may be located. In the case of a call placed through a Gateway Server across the Internet the frequency and cadence detection may be performed by equipment located at the far end point of presence (POP) with that equipment notifying the originating Call Processor of the status of the call. Accordingly, a feature of the present invention is the establishment of acceptable ranges of frequency and cadence attributes of signals from various local telephone company equipment that service the respective candidate telephone numbers. To this end, the system incorporates a method for implementing the CALL PULLBACK mechanism.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete appreciation of the invention and many of the attendant advantages thereof will be readily obtained as the same becomes better understood by reference to the following detailed description when considered in connection with the accompanying drawings, wherein:

Figure 1 is a system level block diagram of a conventional telephony network that includes a Call Processor at a destination facility;

Figure 2 is a timing diagram illustrating an exemplary variation in cadence and frequency of signals provided by different local telephone equipment;

Figure 3 is a flowchart of a method for handling a call in a conventional relay Call Processor;

Figure 4 is a system level block diagram of a Virtual Network having a central Call Processor according to the present invention;

Figure 5 is a block diagram of components in the CALL PULLBACK mechanism according to the present invention;

Figure 6 is a flowchart of a process for contacting an intended recipient by way of the Virtual Network and implementing the CALL PULLBACK mechanism according to the present invention;

Figure 7A is a flowchart of a process for identifying and associating local telephone equipment attributes with candidate customer numbers stored in a computer readable medium according to the present invention;

Figure 7B is a flowchart of a call screening process employed by a Lucent/Octel Node-Overture Call Processor;

Figure 7C is an annotated tone information screen for a failed ring-no-answer; and

Figure 8 is a block diagram of a network of interconnected Virtual Networks that enable both voice and fax messages and other signals to be transported from a source terminal to a destination facility.

BRIEF DESCRIPTION OF THE APPENDIX

An appendix is attached hereto, that contains an application delay table with an index of available delays.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring now to the drawings, wherein like reference numerals designate identical or corresponding parts throughout the several views, Figure 4 is a system level block diagram of a network 400 according to the present invention. A feature of the network 400 is a Virtual Network Call Processor 20 that is separated from the company owned facility 7 located on property owned by an employee of a customer who subscribes to the network 400. The Virtual Network Call Processor may be implemented as a VIRTUAL VOICE NETWORK NODE, offered by TOUCHTONE TECHNOLOGIES Inc. (T3i) and includes a variety of equipment, including a switch, one or more Call Processors with on-board IVR units, multiple T1-spans and or a Gateway Server or Servers that may reside on a network (such as a local area network, LAN) with other equipment. As will become clear, the Virtual Network Call Processor 20 may also operate completely independently of equipment owned and operated by private corporations, and may be used to provide Call Processor, auto attendant, IVR and facsimile functions for individuals with no access to corporate PBX resources. The Virtual Network Call Processor 20 may also be adapted to provide plug-in applications such as Unified Messaging where e-mail may be stored in a mailbox along with voice and fax messages. These e-mail messages may then be read by the voice server to the subscriber. In particular, the Virtual Network Call Processor 20 connects via private or public lines 18 to a source telephone 1. The private or public lines 18 may be part of the PSTN 3, or private lines owned or leased by individual consumers. While the term "lines" is used, these lines may also be wireless links such as private microwave links, or terrestrial or space-based cellular and wireless communication links, or an Internet Backbone employing Gateway Servers for example. Furthermore, the source telephone 1 need not be a conventional telephone, but may also be other communication devices that transmit data from one location to another such as a facsimile device, computer, computer telephone or Internet accessible terminal, for example.

The Virtual Network Call Processor 20 connects to both the source telephone 1, as well as the PSTN or Internet Backbone through a Gateway Server 3, by way of communication links 21, which may be private or leased lines, for example. The source telephone 1, also connects directly to the PSTN 3, which is made up of an interconnected network of equipment owned by companies that service different regions of the United States (or the equivalent of other national and private communication networks in other countries). The PSTN 3, illustrated in Figure 4, includes an interconnected network of three sets of local telephone equipment 30A, 30B and 30C, located in three geographically distinct regions (region 1 - region 3). As previously discussed, the local telephone equipment 30A-30C are often different systems that have different signaling attributes. For example, the telephone equipment 30A may produce a fast busy signal with different signal features than that of local telephone equipment 30C. In particular, as presently recognized, the difference may be in the form of frequency and cadence differences, where "frequency" refers to signal pitch and "cadence" refers to a rhythm of the respective on and off tone cycles that form a beat. Thus, local telephone equipment 30A, which may service a home office 28 may have distinctive frequency and cadence characteristics as compared with that of the local telephone equipment 30B that services the intended recipient's mobile telephone 26, or the local telephone equipment 30C that services the office telephone 11 at the office facility 7. While the PBX 9 at the office facility 7 can receive the phone call directly from source telephone 1, the PBX 9 is capable of only transferring the call internally with devices connected to the PBX 9 or relying on a call forwarding mechanism 5 offered by the local telephone equipment 30C (see, e.g., Figure 1).

However, by subscribing to services offered by the Virtual Network Call Processor 20, the intended recipient is given the option to invoke the CALL PULLBACK mechanism 22 in the Virtual Network Call Processor 20, which, if desired, allows the user to have calls sent to one of any number of candidate locations, each of which may or may not be serviced by different local

telephone equipment. Moreover, because the Virtual Network Call Processor 20 is centrally located (i.e., accessible to parties external to a company's PBX 9), the Virtual Network Call Processor 20 is available for use by many different users, not just users of the PBX 9. Each user can have phone calls that originate at the source telephone 1 be forwarded to the Virtual Network Call Processor 20 and thereby invoke the CALL PULLBACK mechanism 22. The CALL PULLBACK mechanism enables a screened type of call transfer, as compared to a blind transfer where the call is sent to one of several candidate locations without regard for whether the user actually picks up the transferred call at that location.

The Virtual Network Call Processor 20 is shown as part of a node that is made up of a Call Processor, PBX, IVR and other equipment. However, the node may be included as part of a hub, where a hub is one or more digitally networked Call Processors and PBX systems (as will be discussed later in reference to Figure 8).

The process flow for handling a new telephone call is described below, followed by several examples that illustrate how the CALL PULLBACK mechanism 22 is employed. A caller uses a source telephone 1 to attempt to contact an intended recipient at an office telephone 11. The call originating at the source telephone 1 is switched through the PSTN 3 and routed to the PBX 9 at the office facility 7. The PBX 9 then presents the caller with an inquiry, asking the caller to identify an extension for the office telephone 11. In response, the caller enters an extension and the PBX 9 attempts to route the telephone call to the office telephone 11. If all the lines to the PBX 9 are busy, or if there is a ring, but no answer at the destination telephone 11, or all calls are directly forwarded to the Virtual Network Call Processor 20 using the call forward mechanism in the local telephone equipment, the Virtual Network Call Processor 20 receives the call and subsequently processes the call. Alternatively, the call may be transferred directly into the Virtual Network Call Processor 20 by an operator or other company personnel at the office facility 7. As a further alternative, a caller may dial directly into the

Virtual Network Call Processor 20, or be forwarded in by call forwarding previously set up at the customers Central Office 30C under the following conditions:

Ring no answer on the companies main number or numbers.

An example of this usage could be that no one is available to answer, i.e. after hours, weekends, holidays or the company has suffered a catastrophe in which the facility has been destroyed. A Virtual Voice Network utilizing CALL PULLBACK technology may be programmed to allow designated personnel at a company to call into a Node, enter a password and with a few keystrokes have callers processed to telephones other than those at the company such as the home telephones of company personnel. Even if the company is physically gone, business may still be conducted.

Busy on the company's main number or numbers.

All trunks or lines are busy due to traffic or being busied out at the central office while repair or reprogramming work is being performed.

All calls forward with or without ring reminder.

Some companies provide an after hours courtesy to their callers by taking the time to program their main number so that it forwards to a Virtual Voice Network Node without the caller having to listen to a number of rings.

Forwarding with multiple talk paths.

In the case of a customer location with only one trunk or line, one or more callers may reach a Node at the same time when the given line has multiple talk paths.

When the call is transferred to the Virtual Network Call Processor 20, the Virtual

Network Call Processor 20 recognizes the telephone number that the source telephone 1 was attempting to contact by using direct inward dial, (D.I.D.), automatic number identification, (ANI), or direct number identification system, (DNIS). If the telephone number is associated with an office location, the caller is presented with an options menu (described in audible format) asking the caller to select the person or department with whom the caller wishes to speak. Once selected, the caller is placed on soft hold, while the Virtual Network Call Processor 20 dials an external telephone number and initiates a call progress tone detection operation as will be discussed with respect to Figure 5. The CALL PULLBACK mechanism 22 may then consult a list of stored candidate numbers at which the intended recipient may be located, where the numbers stored are provided by the intended recipient when the intended recipient enters (or updates) a user profile, perhaps when the intended recipient originally subscribes for service. Sequentially, the CALL PULLBACK mechanism 22 attempts to contact the intended recipient at the respective destinations (for example home office 28 or mobile phone 26). If the intended recipient is not located at the first candidate location, the call is "pulled back" and if desired by the customer the CALL PULLBACK mechanism 22 informs the caller that it is about to dial the next location as well as offering the caller other options such as leaving a message. If the caller does nothing, the CALL PULLBACK mechanism 22 may consult from memory the next candidate number, and then attempt to contact the intended recipient at that next candidate number. The CALL PULLBACK mechanism 22 may operate in this fashion until all of the candidate locations have been investigated. If the intended recipient has still not been located, the Virtual Network Call Processor 20 allows the caller the options of leaving a message, contacting an operator, or dialing another extension, for example. On the other hand, if the caller is available at one of the candidate locations, the caller remains on soft hold, while the virtual call network processor 20 presents the intended recipient with a call announcement, such as "this is a call for XYZ Engineering Company, press # to accept or * to reject". If the call is accepted,

the calling party and the intended recipient are connected. If the call is rejected, the calling party is informed that the "name" does not answer and is offered further options, such as speaking with an operator, leaving a voice mail message or dialing another extension for example.

While the calling party is placed on soft hold, the CALL PULLBACK mechanism 22, begins a call process tone detection operation, while dialing the external telephone number, as will be discussed with respect to Figure 5.

The connection that is made by the Virtual Network Call Processor 20 may be made to any phone or device that can be dialed directly, even if the call is placed over the Internet, voice over Internet Protocol. Examples of such devices include cell phones (terrestrial and satellite based), direct inward dial (D.I.D.) telephone numbers, business or home telephone numbers, "Multiserve" or similar service telephone numbers, facsimile devices, computers, etc. A feature of the Virtual Network Call Processor 20 is that the transfer of the call from the Virtual Network Call Processor 20 is made to the intended recipient even though the intended recipient is located on a different PBX than the transferring party.

However, the vast majority of the calls processed are to company departments or fixed locations rather then people who are moving from location to location. In such cases, a call is processed to a given telephone and if not answered, the caller is offered the options of leaving a voice mail message, dialing another extension, dialing 0 for the operator or returning to a portion of the menu where another selection can be made.

Figure 5 is a more detailed block diagram of the CALL PULLBACK mechanism 22 shown in Figure 4. The call pullback mechanism 22 includes an application delay adjustment mechanism 24 as shown in Figure 4, the components of which include a random access memory (RAM) 241, read only memory (ROM) 243, hard disk drive (not shown) and internal interface 245 that are connected to a bus 503. An external interface circuit 501 provides the physical interface, and lower level protocol operations for communicating data between the bus 503 and

the private or public lines 18 and 21, which ultimately connect to the source telephone 1 and PSTN 3 as shown in Figure 4. Additional lines may connect to the external interface 501. A processor 505 is a single processor, although multi-processor architectures, as well as hybrid processor and digital signal processor components may be used as well. Additional processors may be included in the CALL PULLBACK mechanism 22, such as in the application delay adjustment mechanism 24, tone detector 507 and frequency and cadence analyzer 509. Operationally, the transferred call comes in through line 21 to the external interface 501, but alternatively, in a direct dial context, the call may come in directly through line 18. Subsequently, the external interface 501 identifies the party to be called, and retrieves a data file associated with the intended recipient, as identified in the call, by way of the bus 503. Part of data file is a first candidate number, which the CALL PULLBACK mechanism 22 will attempt to contact the intended recipient. The intended recipient's call is placed on hold, while the processor 505 initiates another call on an external line (one of the other lines 18-21).

The tone detector 507 is placed on this external line, so as to determine the call progress status of the call made on the external line. The CALL PULLBACK mechanism 22 places the tone detector 507 on the external line so as to determine if a type of busy signal is present. If the busy signal is present, the calling party is removed from hold and the intended recipient's greeting or, alternatively, the intended recipient's name and condition is audibilized to the calling party. At this time, additional options may be offered to the calling party. However, if no busy signal is detected, the CALL PULLBACK mechanism 22 sends out a series of ticking sounds so the called party will know that this is a call from their Virtual Voice Network and if they choose to wait they will hear the announcement of the type of call they are receiving, example, (Sales verses Customer service) and press # to accept, or * to reject before or after the announcement is made.

The frequency and cadence analyzer 509 characterizes different types of signals from the

local telephone equipment serving the candidate location at which the intended recipient is attempting to be located. The signals to be detected include slow busy signals, fast busy signals, ringing signals, answered signals, and ring no answer signals so that additional candidate locations may be searched and/or the caller may be informed of the status of locating the intended recipient. The frequency and cadence analyzer 509 includes a predetermined set of operations that are intended to interpret the various busy signals, ringing signals, answered signals and the like produced at the far end central office. All frequency and cadence analysis is done on the fly by the frequency and cadence analyzer 509 which consults the application delay table for acceptable cadence values.

As previously discussed, the frequency and cadence of different signals varies between local telephone companies equipment. Accordingly, the frequency and cadence analyzer 509 communicates over the bus 503 to receive the application delay parameters from the application delay adjustment mechanism 24. These parameters are included in the RAM 241, although may also be included in the ROM 243 (conveniently implemented as an EEPROM) or at a remote memory accessible by the internal interface 245, by way of the bus 503.

Alternatively, the frequency and cadence analyzer 509 includes an acoustical signature mechanism that compares respective acoustical signatures against a saved set of acoustical signatures saved in the ROM 243, so as to determine if the response received from the local telephone equipment is a slow busy, a fast busy, etc. The frequency and cadence analyzer 509, also incorporates pattern recognition software that attempts to compare and identify signals received from local telephone equipment, on an on-the-fly basis. If no busy signal is detected, the frequency and cadence analyzer 509 sends out a series of ticking sounds and when the call is answered, a call announcement operation is conducted. The ticking sounds are sent out to alert the called party that their Virtual Voice Network is calling and not someone else. During or after call announcing the called party may then chose to accept or reject the call or if more then one

part of the Network directs calls to them they may choose to listen to the full call announcement before accepting or rejecting the call.

The processor 505 includes an internal memory for program storage and holding intermediate calculation results. However, ROM 243 also includes a number of software objects that are invoked by the processor 505, when analyzing and assessing the respective attributes of the local telephone equipment.

Figure 6 is a flowchart of a process flow for contacting an intended recipient by way of the Virtual Network Call Processor 20 as it implements the CALL PULLBACK mechanism. The process begins in step S51, where the calling party dials the number for the intended recipient at a particular number. The process then proceeds to step S53, where an inquiry is made regarding whether the intended recipient answers the phone call (perhaps by way of a local PBX, such as PBX 9 in the office facility 7 of Figure 4). If the intended recipient answers the phone call, the process proceeds to step S55 where the phone call is connected to the intended recipient, and subsequently the process proceeds to step S65 where all the call processing is completed. Alternatively, steps S51, S53 and S55 may be performed by dialing directly the office number of the intended recipient, thereby bypassing the virtual call network processor 20. If the intended recipient is unavailable to answer, the local central office invokes a call forwarding operation that forwards the call directly to the Virtual Network Call Processor.

A majority of the time a given caller reaches a Virtual Voice Network is because the caller was calling a company rather then an individual. Although no two Virtual Voice Networks need to be identical, the majority of them greet the caller and instruct the caller to enter an extension number or choice. Some extensions do not process calls, some only process calls to a single number all of the time and some process calls to multiple numbers. CALL PULLBACK is invoked when a caller is offered other options after placing an unsuccessful call to a telephone not residing at the same location and not directly connected to the T3i Virtual Voice Network.

Call announcing and dialing multiple phone numbers are enhancements that the customer may or may not wish to use.

If the response to the inquiry in step S53 is negative, the process proceeds to step S57 where the call is forwarded (transferred) to the Virtual Network Call Processor 20, where the Virtual Network Call Processor 20 attempts to contact the intended recipient at one of the predetermined numbers stored at the Virtual Network Call Processor 20. In step S57 the caller is placed on soft hold, while an attempt is made to contact the intended recipient by way of an external line. The process then flows to step S59, where an inquiry is made regarding whether the intended recipient answers the call from the Virtual Network Call Processor at a first number stored in the Virtual Network Call Processor 20. Step S59 may have to be repeated if the intended recipient is not located at the first number and additional numbers are included in the intended recipient's profile that may be automatically checked by the Virtual Network Call Processor 20.

If the response to the inquiry in step S59 is affirmative, the Virtual Network Call Processor announces the call to the intended recipient in step S61. By announcing the call, the intended recipient has the option to receive the telephone call from the calling party, or have the Virtual Network Call Processor inform the calling party that the intended recipient is unable to receive the call. By announcing the call to the intended recipient, the intended recipient knows how to answer the phone, for example, when the calling party's call is taken off soft hold, and connected to the intended recipient's telephone. After step S61, the process proceeds to step S65, where call processing is completed and subsequently the process ends. On the other hand, if the response to the inquiry in step S59 is negative, the Virtual Network Call Processor invokes the CALL PULLBACK mechanism where the call remains on soft hold, and an external line (either the same external line as before, or another line) is used to attempt to contact the intended recipient at the next number identified in the profile of the intended recipient. Using this

example, the CALL PULLBACK mechanism would remove the caller from soft hold and announce that the called party was unavailable and offer further options, one of which could be to dial another number. This process of tracking-down the intended recipient proceeds until all of the candidate locations have been exhausted, at which time the Virtual Network Call Processor either takes a voice mail message, asks the calling party if they would like to identify another person to whom to route the call, etc. Of course, if the CALL PULLBACK mechanism successfully contacts the intended recipient, and the intended recipient decides to receive the call, the caller is then taken off hold, and connected to the intended recipient. Subsequently, the process is completed in step S65 and the process ends.

Application delays are timing values set in the Call Processor portion of the Node. Delays described in the Appendix are used to detect critical tone cadences that the Central Office provides to the Node equipment. These cadences indicate specific call conditions such as a ringback tone indicates that a called number is ringing, and a busy indicates that the called party is busy.

Cadence values are normally set by selecting a PBX type and making modifications to the equipment as needed. As it is not known what PBX type a given Central Office would have or what effect the state of repair or software level would have on the cadences provided, a starting point is to choose the PBX closest to the one which was part of a first node implemented by a user of the system.

After adjusting the Call Processor's cadence recognition to the first Central Office PBX, new adjustments can be made as Central Offices are added and tested to make sure that the new adjustments work with previous Central Office PBXs'. Fail-safe mechanisms are included as features of the CALL PULLBACK to catch any caller that hit an unexpected cadence. These fail-safe mechanisms allow callers the options of reaching a live operator or leaving a message as well as providing first hand intelligence regarding what happened and where an unexpected event

occurred. As the system matures, the fail-safes are not needed as frequently because the system's attributes will become more completely characterized with time.

Problems chiefly occur in areas where the Call Processor detects an answered condition while monitoring a single interrupted-ringback and with slow-busy and fast-busy cadences. When a call is screened, the equipment looks for acceptable cadences for a single interrupted ringback, slow-busy, fast-busy or that the call has been answered. To process a transfer application-delay, indexes are referenced that show the maximum and minimum ON/OFF periods for any tone. If the tone cadence detected does not comply with the ranges set for single ring back, slow busy or fast busy, the Call Processor determines that the call has been answered and the call transfer is completed.

In the case of dead air, such as the Central Office dropping the call, the caller would be removed from soft hold and a fail-safe mechanism would take over. One problem that occurs is when a Central Office recording is played such as an all circuits are busy or that the person being called is out of the area or unavailable. The Call Processor detects an answer and sound, such as someone speaking, and completes the transfer. These occurrences can be kept to a minimum and caller frustration reduced by the following methods:

Prior testing and identification - this allows the network designer to inform the customer of a potential problem and make needed changes or record a special greeting so that the caller has the opportunity to record a message or go to an operator before the called party's number is dialed;

Pulling the call back before the far end recording is played; and

Most customers and callers are used to the recordings being played and are not troubled by them. If the tone cadences are within acceptable ranges, call screening by the called party may be employed and the call accepted or rejected.

Figure 7A is a flowchart of a process for identifying local telephone equipment attributes,

such as frequency and cadence. The process begins in step S661 where the called party is dialed by a node. Subsequently, the process proceeds to step S663, where the cadence and frequency information from signals produced by the local telephone equipment is observed by the node. When observing the cadence and frequency information, the cadence and frequency information is characterized for subsequent processing. The process then proceeds to step S665, where application delays are identified that correspond with the frequency and cadence information that was characterized in step S663. The process then proceeds to step S667, where the node takes appropriate action for the call based on predefined custom parameters and/or reacts to the cadence events, where the reaction is a function of detecting which signals are in fact produced by the local telephone equipment. Subsequently, the process ends.

Figure 7B is a flow chart of a process used by a NODE-OVERTURE Call Processor to screen calls. The process begins in S71, where the call is transferred and the NODE-OVERTURE Call Processor dials the called number and begins looking at tone patterns received from the local telephone equipment. Subsequently, the process proceeds to step S72, where an inquiry is made regarding whether the tones that are received comply with the ranges set by delays 49, 50, 51 or 52, as identified in the appendix attached hereto. If the response to the inquiry in step S72 is negative, the process proceeds to step S73, where the call is considered to have been answered, and subsequently the process ends. However, if the response to the inquiry in step S72 is affirmative, the process proceeds to a ring back inquiry in step S74, where an inquiry is made regarding whether the received tones comply with the ranges set by delays 53, 54, 55 or 56. If the response to the inquiry in step S74 is affirmative, the phone rings and a ring back is monitored. The ring back minimum and maximum tone off period are included in the appendix. The process then concludes.

However, if the response to the inquiry in step S74 is negative, the process proceeds to step S76, where a slow busy inquiry is made. The inquiry in step S76 inquires whether the tones

comply with the ranges set by delays 69, 70, 71 or 72. If the response to the inquiry in step S76 is affirmative, the process proceeds to step S77, where the call is pulled back and the NODE OVERTURE Call Processor speaks the name and condition or greeting and subsequently the process ends. However, if the response to the inquiry in step S76 is negative, the process proceeds to the fast busy inquiry in step S78, and an inquiry is made regarding whether the tones comply with the ranges set by delays 73, 74, 75 or 76. If the response to the inquiry in step S78 is affirmative, the process proceeds to step S79, where the call is pulled back, and an indication is spoken indicating that the call is "invalid" and then the process ends. However, if the response to the inquiry in step S78 is negative, the call is answered in step S80 and then the process ends.

Figure 7C is an exemplary display of information that would be displayed on a monitor screen for a situation where a ring-no-answer operation fails. Application delays that produce a failure are first tested by assigning a mailbox that dials the problem telephone number to a special area code specific object. The port specific print tone trace is activated and the node's Call Processor is called through the port in question. The port will have a speakerphone butt set placed on it so that there is an audible awareness on the part of the person performing the task so as to determine what events are in fact occurring. When answered, the test mailbox is dialed and the tone on and tone off events are monitored in milliseconds as shown on a computer screen. In Figure 7C, a display 701, which, may either be a simultaneous display, or printout or otherwise of a stream of code, illustrates tone information that is displayed for a ring-no-answer operation that fails. As shown in annotation 702, the caller enters DTMF digits, which are the digits associated with the phone to be contacted. In annotation 703, the NODE OVERTURE monitors the line for a tone associated with a dial tone and then subsequently detects the presence of the dial tone. Once the dial tone is detected, and annotation 704 shows that the NODE OVERTURE dials the number associated with the DTMF digits. Annotation 705 indicates that the NODE OVERTURE ignores the first change in tone for a predetermined period of time. Subsequently,

as indicated in annotation 706, the NODE OVERTURE monitors call progress tones from the PBX so as to determine the status of the called extension. Finally, in annotation 707 a failure is indicated when the NODE OVERTURE detects an answer condition because one of the tones of the PBX does not conform to the delays in the application delay table, listed in the appendix.

The tone values in the print tone trace of Figure 7C may be modified to the correct values by using the appropriate commands. For example, in the case of Figure 7C, the application-delay indexes that refer to the error received are indexes 50 and 54, included in the appendix. Note that the failure occurred when the PBX sent a TONE ON for 790 ms. The NODE OVERTURE was set to expect a TONE ON (ring back) for no less than 800 ms and no greater than 1200 ms. This range between 800 ms and 1200 ms is referred to as the "window". In Figure 7C, the window for the silence period (TONE OFF) between adjacent rings is set to no less than 2800 ms and no greater than 3400 ms. The TONE OFF values are within that window.

Figure 8 is a block diagram of an intelligent network of Virtual Network Call Processors (20A, 20B, 20C, 20D) connected together, as shown, in a ring configuration, although other configurations may be performed as well, such as a star or non-geometric specific interconnected configuration. Each of the Virtual Network Call Processors 20A through 20D is configured as several interconnected nodes, a node being a PBX having a Call Processor. The Virtual Network Call Processors 20A-20D are the respective hubs, and thus each serves a different geographical area. As a consequence, a source communication device 1A may connect to the Virtual Network Call Processor 20A (or other Virtual Network Call Processor 20B-20D) by way of the local telephone equipment 30A, or directly to the Virtual Network Call Processor 20A. The Virtual Network Call Processor 20A may then route the information (voice, or other type of data, such as image data, facsimile data, etc.) via the other Virtual Network Call Processors (20B-20D) and then to a destination facility 7 either directly from Virtual Network Call Processor 20D, or via the local telephone equipment 30B.

Because the respective hubs 20A-20D are digitally linked, via dedicated point to point connections or by the use of VPNs' (Virtual Private Networks) or through a gateway server over the Internet, no charges for long distance services are required, although the network of Virtual Network Call Processors 700 certainly could charge a fee for such services or other fee-based links may be used as well.

An example application of the network architecture of Figure 8, might be if either an e-mail message, a facsimile message or other message such as a digitized voice, video or data file were intended to be left with a person in Florida (serviced at Virtual Network Call Processor 20A), a copy of that message may be routed through the network of hubs 20A-20D and to the destination facility 7. Since the connections are by way of dedicated links or VPNs, there are no long distance charges. In addition to the data relaying service, each of the respective Virtual Network Call Processors 20A-20D may also provide the standard call processing features described in Figures 4-5, for example.

The inventive system may include a CALL PULLBACK mechanism that employs a primary rate interface (PRI) that is compatible with National ISDN standards deployment of Simplified Message Desk Interface (SMDI).

The mechanisms and processes set forth in the present description may be implemented using one or more general purpose microprocessors programmed according to the teachings of the present specification, as will be appreciated to those skilled in the relevant art(s). Appropriate software coding can readily be prepared by skilled programmers based on the teachings of the present disclosure, as will be apparent to those skilled in the relevant art(s). The present invention thus also includes a computer-based product which may be hosted on a storage medium and include instructions that can be used to program a computer to perform a process in accordance with the present invention. The storage medium can include, but is not limited to, any type of disk including floppy disk, optical disk, CDROM, magneto-optical disk, ROMs,

RAMs, EPROMs, EEPROMs, flash memory, magnetic or optical cards, or any type of media suitable for storing electronic instructions, either locally or remotely.

Obviously, numerous modifications and variations of the present invention are possible in light of the above teachings. It is therefore to be understood that within the scope of the appended claims, the invention may be practiced otherwise than as specifically described herein.

CLAIMS:

1. A virtual network call processing system, comprising:

a communication line interface configured to be connected to a source terminal and receive a calling message from the source terminal directed to an intended recipient;

a call processor with a call pullback mechanism including,

a data processor, and

a computer readable memory having computer readable instructions encoded therein that when executed by said data processor implement a local telephone equipment characterization mechanism that characterizes signaling attributes of signals produced by local telephone equipment that service different geographical locations at which the intended recipient may be located; and

a signal determination mechanism configured to determine whether the signals provided by the local telephone equipment have at least one of a frequency and cadence associated with a signal event that includes at least one of a fast busy signal, slow busy signal, ringing signal, answered signal, and ring-no-answer signal.

2. The system according to Claim 1, further comprising:

an error handling mechanism configured to process the calling message when the signal determination mechanism fails to determine that the signal event occurred.

3. The system according to Claim 1, wherein:

said signal determination mechanism includes a software tool programmed to recognize the at least one of the frequency and cadence associated with the signal event from signals associated with the local telephone equipment.

4. The system of Claim 1, wherein:

said computer readable memory includes

an intended recipient profile, having a first destination number and a second destination number, and

said call pullback mechanism further includes a recipient contact mechanism being configured to attempt to first contact said intended recipient via an external line at the first destination number, and if not present, being configured to attempt to contact said intended recipient at the second destination number.

5. The system of Claim 4, wherein:

said signal determination mechanism includes a tone detector being configured to detect when said calling message on said communication line is answered, or said signal event occurs; and

said system further includes a call announcing mechanism configured to execute a whisper transfer to the called party that allows the called party to accept or reject the calling message.

6. The system of Claim 5, wherein:

said tone detector being configured to notify the recipient contact mechanism when said communication line is not answered so said recipient contact mechanism proceeds to contact said intended recipient at said second destination number.

7. The system of Claim 1, wherein:

said local telephone equipment characterization mechanism includes a frequency characterization mechanism, configured to characterize a frequency of the signals produced by

respective of said local telephone equipment.

8. The system of Claim 1, wherein:

said local telephone equipment characterization mechanism includes a cadence analyzer configured to analyze a cadence of respective of the signals provided by respective of said local telephone equipment.

9. The system of Claim 8, wherein:

said local telephone equipment characterization mechanism includes a frequency analyzer configured to analyze a frequency of the signals provided by respective of said local telephone equipment.

10. The system of Claim 1, further comprising:

a signal feature normalization mechanism, including an application delay adjustment mechanism configured to adjust respective application delays in said call processor so as to standardize signal attributes provided by respective of the local telephone equipment.

11. The system of Claim 10, further comprising:

another data processor and another computer readable memory configured to implement another local telephone equipment characterization mechanism and another signal feature normalization mechanism, said processor and computer readable medium being connected to said another processor and said another computer readable medium by an intercity communication link.

12. A method for processing a call in a virtual network call processing system,

comprising the steps of:

receiving a calling message from a source terminal directed to an intended recipient;

retrieving a data profile of the intended recipient from a computer readable medium;

identifying a number to contact the intended recipient via a local telephone equipment;

characterizing signal attributes of signals provided by the local telephone equipment;

initiating the call on an external line with said number at said local telephone equipment;

normalizing the signal from the local telephone equipment;

transferring the calling message if the call is accepted by the intended recipient, but retaining the calling message for future processing if the call is not accepted by the intended recipient.

13. The method of Claim 12, further comprising the steps of:

identifying another number in said data profile of said intended recipient;

calling on at least one of the external line and another external line, said another number at another local telephone equipment;

normalizing a signal from the another local telephone equipment;

transferring the calling message if accepted by the intended recipient at the another number, but retaining the calling message for further processing if not accepted.

14. The method of Claim 13, further comprising the step of:

producing a tone on said external line and detecting when said external line is answered.

15. The method of Claim 12, wherein:

said characterizing step comprises characterizing a frequency of the signal provided by the local telephone equipment.

16. The method of Claim 12, wherein:

said characterizing step comprises characterizing a cadence of the signal provided by the local telephone equipment.

17. The method of Claim 16, wherein:

said characterizing step further comprises characterizing a frequency of the signal provided by the local telephone equipment.

18. The method of Claim 12, wherein:

said normalizing step comprises adjusting respective application delays so as to standardize signal attributes of the signal from the local telephone equipment.

19. The method of Claim 12, wherein:

said calling step comprises passing the calling message from a first hub to a second hub, prior to reaching the local telephone equipment.

20. A computer readable medium encoded with computer readable instructions for use in a system having a communication line interface configured to be connected to a source terminal and configured to receive a calling message from the source terminal directed to an intended recipient, said computer readable instructions when executed by a data processor implement a system comprising:

a local telephone equipment characterization mechanism that characterizes signaling attributes of signals produced by local telephone equipment that service different geographical locations at which the intended recipient is located;

a signal determination mechanism configured to determine whether the signals provided

by the local telephone equipment have at least one of a frequency and cadence associated with a signal event that includes at least one of a fast busy signal, slow busy signal, ringing signal, answered signal, and ring-no-answer signal; and

a call pullback mechanism configured to call an intended recipient and transfer a calling message to said intended recipient if said signal determination mechanism determines said call is answered, but not transferring said calling message if said signal determination mechanism determines that said call is not answered.

21. A virtual network call processing system, comprising:

means for receiving a calling message from a source terminal directed to an intended recipient;

means for identifying a number to contact the intended recipient via a local telephone equipment;

means for characterizing signal attributes of signals provided by the local telephone equipment;

means for initiating a call on an external line with said number at said local telephone equipment;

means for normalizing the signal from the local telephone equipment; and

means for transferring the calling message if the call is accepted by the intended recipient, but retaining the calling message for future processing if the call is not accepted by the intended recipient.

CALL PROCESSING SYSTEM, METHOD AND
COMPUTER PROGRAM PRODUCT

ABSTRACT OF THE DISCLOSURE

A system, method and computer program product implement a Virtual Network Call Processor with a CALL PULLBACK mechanism for providing a type of screened call transfer. Callers, while attempting to contact an intended recipient, have their calls sent to the Virtual Network Call Processor, which places the caller on soft hold while attempting to locate the intended recipient. The Call Processor uses another external line to call the intended recipient at one of a number of predetermined locations identified by stored numbers where each number is serviced by perhaps different local telephone equipment having different characteristics and attributes. The CALL PULLBACK mechanism is used to identify signaling attributes of signals provided by the respective local telephone equipment, by analyzing frequency and cadence information from the signals and normalize the signals so as to detect a status of the Call Processor's attempt to reach the intended recipient. The signaling attributes and customer-specific information are controlled by objects, which are well thought out preprogrammed and proven software constructs that simplify programming and ensure reliable operations. The calling party is kept on soft hold while the intended recipient of the call is attempted to be contacted at the different locations. If the CALL PULLBACK mechanism determines that the signals provided by the local telephone equipment, after being normalized, indicate the intended recipient does not pick up the call, the CALL PULLBACK mechanism attempts to reach the intended recipient at another one of the numbers, all the while the calling party is kept on soft hold. In this way, the global Virtual Network Call Processor, is capable of servicing not only individuals and companies serviced by a single PBX with a call process, but also for any number of other users not serviced by the PBX.

APPENDIX

APPLICATION DELAY TABLE.

INDEX	DELAY (msec.)
0	0
1	7000
2	5000
3	500
4	5000
5	1200
6	1000
7	30000
8	800
9	4000
10	6000
11	1000
12	1000
13	10000
14	20000
15	25000
16	2000
17	20
18	1000
19	200
20	240
21	100
22	140
23	200
24	260
25	300
26	400
27	260
28	460
29	800
30	900
31	1200
32	1500

33	2000	
34	2100	
35	2700	
36	3900	
37	4000	
38	6000	
39	1000	
40	4000	
41	10000	
42	1500	
43	1500	
44	2000	
45	500	
46	1700	
47	700	
48	500	
49	2500	Maximum tone on period for any tone. When ringback, busy or fast busy is encountered this delay is used to determine whether the tone on is a valid tone. When an encountered tone on is longer then this delay it is assumed to be an answer.
50	50	
51	4600	Maximum tone off period for any tone. When ringback, busy or fast busy is encountered this delay is used to determine whether the tone off is of valid duration. When tone off is greater then this value it is assumed to be an answer.
52	40	
53	2500	Maximum tone on period. This is the longest tone on event that will be considered as ringback. This application delay is used to determine whether the tone on cadence event being monitored is ringback. If the tone on event is longer then this application delay it is assumed to not be ringback.

54	800	
55	4600	Ringback maximum tone off period. This is the longest tone off event that will be considered as ringback. If the tone off event being monitored is shorter then this application delay it is assumed to not be ringback.
56	2000	Ringback minimum tone off period. This is the shortest tone off event to qualify as ringback. This application delay is used to determine whether the tone cadence being monitored is ringback. If the tone cadence being monitored is less in duration then this value it is assumed to not be ringback
57	2300	
58	20	
59	15000	
60	20	
61	500	
62	300	
63	500	
64	300	
65	3200	
66	2800	
67	280	
68	120	
69	600	
70	400	
71	600	
72	400	
73	320	
74	180	
75	320	
76	180	

The following is the issued U.S. Patent Serial No. 6,088,437, dated July 11, 2000, referred to herein as "CALL PULL-BACK."

8006-0006-52

TITLE OF THE INVENTION

CALL PROCESSING SYSTEM, METHOD AND COMPUTER PROGRAM PRODUCT

CROSS REFERENCE TO RELATED APPLICATIONS

The present document claims the benefit of the earlier filing date of, and contains subject matter related to that disclosed in, co-pending U.S. provisional application Serial No. 60/082,730 filed April 23, 1998, having common inventorship, the entire contents of which being incorporated herein by reference.

BACKGROUND OF THE INVENTION

Field of the Invention:

The present invention pertains to call processing systems, methods and computer-based products used for telephony systems in which calls may be screened by a called party prior to connection. More particularly, the present invention is directed to voice and data systems that include Call Processors and Gateway Servers among other data communication resources, in which calls targeted for a predetermined location are directed by the programming of a Virtual Voice Network. This direction can be to any device that can be directly dialed, such as a telephone including cell phone, fax machine or modem even if the call for the targeted device is processed across the Internet, (voice or fax over Internet Protocol.)

Discussion of the Background

Advances in modern electronics and digital communication enable individuals to communicate virtually anywhere around the world. With the advent of cellular telephones,

personal communication services, and satellite telephony, individuals in advanced as well as developing societies have an expectation of being able to communicate with others anytime and anywhere in a smooth and seamless fashion. The bulk of existing communication infrastructure is provided by local, regional and long distance telephone companies, i.e., the public switch telephone network (PSTN), which uses land lines, among other resources, for providing point-to-point communications, where each point is identifiable by a separate telephone number. For example, a caller may use a first telephone number when attempting to reach a person at the person's home, but uses a second number for contacting the person at the person's office.

A recent challenge has been how to use the PSTN, components of which contain old technology, to provide the flexibility to support people who want to remain accessible while being mobile. In light of this backdrop, some corporations use private branch exchanges (PBX) at the corporation facilities to provide "smart" functions for handling incoming phone calls (described generically here, but referring to both voice and data calls) for the convenience of its customers. Using these functions, even if an employee is not available, a properly equipped PBX enables an outside caller to be conveniently patched into voice mail, routed to another number, or perhaps transferred to a different facility in an attempt to handle the call in a user-friendly environment.

Figure 1 is a block diagram of a conventional PSTN and PBX based system that enables a source telephone 1 to communicate with a destination telephone 11, of an intended recipient. In Figure 1, the source telephone 1 connects to the PSTN 3 via a line (wired or wireless). The PSTN 3 recognizes the telephone number input at the source telephone 1 and provides the switch infrastructure to ultimately connect the caller with the PBX 9, which has the burden of providing the "operator" interface functions. In many cases, the corporation facilities 7 may incorporate a relay Call Processor with auto attendant functions 13 even though unit costs for such devices could exceed 1.5 million dollars in 1998. An example of such a Call Processor is an

OVERTURE 300 sold by the Lucent/Octel Messaging Division.

The relay Call Processor with auto attendant 13 operates when the call is received by the PBX 9 and attempts to ring the destination telephone 11, while placing the caller on hold or utilizing any of a number of types of integration depending on the makes and models of the equipment. If the destination telephone 11 is not picked-up after a predetermined number of rings, the Call Processor with auto attendant 13 initially reports a message to the caller such as "thank you for calling company A. John Doe is on the telephone so please leave a message, dial another extension, or dial 0 for the operator." The Call Processor with auto attendant 13 is able to handle the call for the employee in this way because the PBX 9 receives and routes all telephone calls within the company facility 7 without having to interface with a variety of different local telephone equipment, each having unique signaling attributes.

For users that do not have the benefit of a corporation's PBX 9, the PSTN offers users a call forwarding operator 5 that, at the instruction of the intended recipient, forwards incoming calls to a secondary number when the intended recipient is unavailable at a primary number. This call forwarding mechanism however employs equipment at the PSTN and does not offer the same degree of convenient voice mail and auto attendant functions offered by the PBX 9 at the company facility 7.

As presently recognized by the inventor, the PBX 9 is an inherently "local" device hosted at a certain destination facility, such as a company. Available for equipment of such expense, smaller devices such as the relay Call Processor with auto attendant 13 are included with the PBX 9 to provide added functionality. Adding to the expense, the relay Call Processor with auto attendant 13 must be customized by technicians when installed at the company facility 7 so as to be compatible with the local telephone company equipment if any screened transfer types of calls were to be placed to external telephone numbers. Customization is needed because the PSTN 3 is not homogenous, but rather made up of numerous equipment of local telephone companies that

may or may not have the same equipment. As an example of different signaling attributes of signals provided by typical telephone equipment, the frequency and cadence of slow-busy signals (or other signals, as will be discussed) may be substantially different from one local telephone to the next. Similarly, other signals such as a fast busy signal, indicating an error is present, differs as well.

Figure 2 is a timing diagram of a ring/silence signal offered by exemplary local telephone company equipment. A high voltage level indicates a ring interval, while a lower voltage indicates a silence interval. For illustrative purposes, the interval "A" may typically range between a maximum tone-on (i.e., ring interval) of 1,200 ms to a minimum of 800 ms, while a typical number may be 1000 ms. The interval "B"(a silence interval) may range between 3500 ms and 2801 ms, with a typical number being 2881 ms. Interval "C" may typically range between 1200 ms and 800 ms, with a typical number being 942 ms. Interval "D" may typically range between 3485 ms and 2899 ms, with a typical number being 2910 ms. Similarly, the interval "E" may typically range between 1200 and 800 ms with a typical time 785 ms (which is less than the stated lower end of the "typical" range, but included to show that it is nonetheless a possibility). Due to this variation in cadence and frequency between signals provided by local telephone equipment, generic relay Call Processors with auto attendant functions are conventionally believed to require the use of technicians to personally customize the "application delays". This approach essentially normalizes the cadence and frequency terms so that the Call Processor can effectively interface with that particular local telephone equipment. Consequently, according to conventional wisdom, it is not believed wise, nor even possible, to use a relay Call Processor with auto attendant function in a central location that operates with different local telephone equipment because the diversity of telephone equipment does not permit the relay Call Processor with auto attendant to handle common signals in a like fashion.

Figure 3 is a flowchart of an example method of how a caller at a source telephone 1 (Fig.

1) attempts to communicate with an intended recipient at the company facility 7. The process begins in step S1, where the caller initiates a call to the intended recipient by dialing a phone number of the company where the intended recipient is believed to be located. The process then proceeds to step S3, where the call is answered by the PBX 9 at the company facility 7, and the PBX 9 passes the call to the Call Processor with auto attendant 13. The process then proceeds to step S5, where the caller is requested to dial the extension of the intended recipient. The process then proceeds to step S7 where an inquiry is made regarding whether the individual identified at that extension is available. While making the inquiry, the Call Processor in the PBX 9 places the caller on hold and rings the destination telephone a predetermined number of times. If the intended recipient does not answer the telephone call after the predetermined number of times or if a busy signal is received, the Call Processor concludes that the intended recipient is unavailable. If the response to the inquiry in step S7 is affirmative, the PBX 9 connects the caller with the intended recipient in step S9 and the process then proceeds to step S11 where the call is completed and then the communication session ends. However, if the response to the inquiry in step S7 is negative, the process proceeds to step S13 where the relay Call Processor with auto attendant 13, audibly presents a set of options to the caller. Typical options include leaving a voice mail message, hitting zero to dial an operator or entering the extension of another party. Once the options are presented, the process proceeds to S15 where the caller selects an option and then in step S17 the selected option is executed. Subsequently the process ends. The Call Processor may also offer other options to the caller, such as attempting to contact the intended recipient at another location. If the caller chooses this option, the Call Processor with auto attendant 13 performs a blind transfer to that other location. Since the Call Processor with auto attendant 13 performs the blind transfer, the Call Processor performs no additional processing of the call even if the intended recipient is not available at the other location. The blind transfer will be made regardless; even if the called party is busy, ring no answer, error tone or dead air.

Some coverage methods employ various call forwarding schemes in the event a called device is busy, or ring no answer. These methods are designed to forward the caller to a receptor mailbox. Often the receptor mailbox is located where the call originated and the called party pays the bill for any forwarding or long distance charges.

As identified by the present inventor, a limitation with conventional devices and methods is that the functions offered by the Call Processor in the PBX 9 are prohibitively expensive for the "small user". In other words, the call processing functions available at the company facility 7 are expensive to purchase and install, and thus are unsuitable for private use. Furthermore, due to differences between the different types of local telephone company equipment employed throughout the PSTN, making a conventional relay Call Processor with auto attendant available to users across a number of different local telephone company equipment is not believed to be possible, due to the different signaling attributes of the equipment employed by the different local telephone companies.

As presently recognized, the installation procedures of PBX 9 with the relay Call Processor are complex in that "hands on" customization and testing of the local telephone equipment is believed to be required in conventional systems when adjusting the destination delays for the relay Call Processor. Such difficulties are factors that contribute to the expense of purchasing and maintaining a Call Processor, even though conventional Call Processors are used over a specific geographical region sharing a common set of telephone equipment.

More centralized functions, such as call forwarding operations provided by the PSTN are incapable of detecting whether a person is available at one of the candidate locations, and "pulling back" the call for further processing if the person is unavailable. Moreover, the call forwarding operations perform a blind transfer of the call, and do not wait to determine whether the destination party will in fact receive the call. As a consequence, the user-friendliness of the call forwarding operation is presently viewed as being sub-optimal.

U.S. Patent No. 5,375,161 describes a telephone control system with branch routing, which includes a call conferencing feature (see, e.g., Figure 14', step 1419) that waits to determine whether or not a user may be located at another number. This technique thus employs precision busy/ring detection that requires *a priori* knowledge of the attributes of the local telephone communication equipment. Without this knowledge, it would not be possible for such a device to operate without significant customization of PSTN equipment at varying locations. Furthermore, the precision busy/ring would not be able to recognize an error tone, which has the same frequency as a busy but a different cadence, because the precision busy/ring unit monitors only frequency, not cadence. Cadence is a variable that fluctuates most from Central Office to Central Office. When considering a conference feature with voice, a number of conditions should be taken into consideration, including hardware sensitivity and ability to be configured, susceptibility to talk off in which the human voice emulates touch tone, and background noise. Regarding background noise, a PC modem, for example, connecting to a service provider for Internet access can cause any touch-tone activated equipment to do unexpected things. Thus, conferencing features are suboptimal.

SUMMARY OF THE INVENTION

Accordingly, a feature of the present invention is to provide a novel system, method and computer based product that overcomes the limitations of the conventional methods and systems discussed above. While a full description of the invention and its various features are described in the following section, a brief, non-exhaustive description of features of the present invention is now described. A facet of this invention is that the Call Processors used normally do not reside at the same location and are not directly connected to a customer's PBX or to the customer's Central Office. If the called party is busy / no answer or an error tone is encountered, the calling party is informed of the status of the called party and may be offered further options.

A "CALL PULLBACK" mechanism is included in a central location (i.e., accessible to geographically separated users) in a Call Processor, which is a component of a virtual Call Processor network. The Call Processor in the virtual Call Processor network places a caller on "soft hold" while attempting to contact the intended recipient at one of various predetermined numbers. In order to overcome the incompatibility issue of operating with different local telephone equipment, a feature of the present invention is a frequency and cadence detection mechanism that is able to detect different characteristics of slow busy, fast busy, ringing, answered, and ring no answer tones as provided by different local telephone equipment. To this end, the Call Processor of the present invention associates different frequencies and cadences with various events occurring with candidate numbers at which the intended recipient may be located. In the case of a call placed through a Gateway Server across the Internet the frequency and cadence detection may be performed by equipment located at the far end point of presence (POP) with that equipment notifying the originating Call Processor of the status of the call. Accordingly, a feature of the present invention is the establishment of acceptable ranges of frequency and cadence attributes of signals from various local telephone company equipment that service the respective candidate telephone numbers. To this end, the system incorporates a method for implementing the CALL PULLBACK mechanism.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete appreciation of the invention and many of the attendant advantages thereof will be readily obtained as the same becomes better understood by reference to the following detailed description when considered in connection with the accompanying drawings, wherein:

Figure 1 is a system level block diagram of a conventional telephony network that includes a Call Processor at a destination facility;

Figure 2 is a timing diagram illustrating an exemplary variation in cadence and frequency of signals provided by different local telephone equipment;

Figure 3 is a flowchart of a method for handling a call in a conventional relay Call Processor;

Figure 4 is a system level block diagram of a Virtual Nctwork having a central Call Processor according to the present invention;

Figure 5 is a block diagram of components in the CALL PULLBACK mechanism according to the present invention;

Figure 6 is a flowchart of a process for contacting an intended recipient by way of the Virtual Network and implementing the CALL PULLBACK mechanism according to the present invention;

Figure 7A is a flowchart of a process for identifying and associating local telephone equipment attributes with candidate customer numbers stored in a computer readable medium according to the present invention;

Figure 7B is a flowchart of a call screening process employed by a Lucent/Octel Node-Overture Call Processor;

Figure 7C is an annotated tone information screen for a failed ring-no-answer; and

Figure 8 is a block diagram of a network of interconnected Virtual Networks that enable both voice and fax messages and other signals to be transported from a source terminal to a destination facility.

BRIEF DESCRIPTION OF THE APPENDIX

An appendix is attached hereto, that contains an application delay table with an index of available delays.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring now to the drawings, wherein like reference numerals designate identical or corresponding parts throughout the several views, Figure 4 is a system level block diagram of a network 400 according to the present invention. A feature of the network 400 is a Virtual Network Call Processor 20 that is separated from the company owned facility 7 located on property owned by an employee of a customer who subscribes to the network 400. The Virtual Network Call Processor may be implemented as a VIRTUAL VOICE NETWORK NODE, offered by TOUCHTONE TECHNOLOGIES Inc. (T3i) and includes a variety of equipment, including a switch, one or more Call Processors with on-board IVR units, multiple T1-spans and or a Gateway Server or Servers that may reside on a network (such as a local area network, LAN) with other equipment. As will become clear, the Virtual Network Call Processor 20 may also operate completely independently of equipment owned and operated by private corporations, and may be used to provide Call Processor, auto attendant, IVR and facsimile functions for individuals with no access to corporate PBX resources. The Virtual Network Call Processor 20 may also be adapted to provide plug-in applications such as Unified Messaging where e-mail may be stored in a mailbox along with voice and fax messages. These e-mail messages may then be read by the voice server to the subscriber. In particular, the Virtual Network Call Processor 20 connects via private or public lines 18 to a source telephone 1. The private or public lines 18 may be part of the PSTN 3, or private lines owned or leased by individual consumers. While the term "lines" is used, these lines may also be wireless links such as private microwave links, or terrestrial or space-based cellular and wireless communication links, or an Internet Backbone employing Gateway Servers for example. Furthermore, the source telephone 1 need not be a conventional telephone, but may also be other communication devices that transmit data from one location to another such as a facsimile device, computer, computer telephone or Internet accessible terminal, for example.

The Virtual Network Call Processor 20 connects to both the source telephone 1, as well as the PSTN or Internet Backbone through a Gateway Server 3, by way of communication links 21, which may be private or leased lines, for example. The source telephone 1, also connects directly to the PSTN 3, which is made up of an interconnected network of equipment owned by companies that service different regions of the United States (or the equivalent of other national and private communication networks in other countries). The PSTN 3, illustrated in Figure 4, includes an interconnected network of three sets of local telephone equipment 30A, 30B and 30C, located in three geographically distinct regions (region 1 - region 3). As previously discussed, the local telephone equipment 30A-30C are often different systems that have different signaling attributes. For example, the telephone equipment 30A may produce a fast busy signal with different signal features than that of local telephone equipment 30C. In particular, as presently recognized, the difference may be in the form of frequency and cadence differences, where "frequency" refers to signal pitch and "cadence" refers to a rhythm of the respective on and off tone cycles that form a beat. Thus, local telephone equipment 30A, which may service a home office 28 may have distinctive frequency and cadence characteristics as compared with that of the local telephone equipment 30B that services the intended recipient's mobile telephone 26, or the local telephone equipment 30C that services the office telephone 11 at the office facility 7. While the PBX 9 at the office facility 7 can receive the phone call directly from source telephone 1, the PBX 9 is capable of only transferring the call internally with devices connected to the PBX 9 or relying on a call forwarding mechanism 5 offered by the local telephone equipment 30C (see, e.g., Figure 1).

However, by subscribing to services offered by the Virtual Network Call Processor 20, the intended recipient is given the option to invoke the CALL PULLBACK mechanism 22 in the Virtual Network Call Processor 20, which, if desired, allows the user to have calls sent to one of any number of candidate locations, each of which may or may not be serviced by different local

telephone equipment. Moreover, because the Virtual Network Call Processor 20 is centrally located (i.e., accessible to parties external to a company's PBX 9), the Virtual Network Call Processor 20 is available for use by many different users, not just users of the PBX 9. Each user can have phone calls that originate at the source telephone 1 be forwarded to the Virtual Network Call Processor 20 and thereby invoke the CALL PULLBACK mechanism 22. The CALL PULLBACK mechanism enables a screened type of call transfer, as compared to a blind transfer where the call is sent to one of several candidate locations without regard for whether the user actually picks up the transferred call at that location.

The Virtual Network Call Processor 20 is shown as part of a node that is made up of a Call Processor, PBX, IVR and other equipment. However, the node may be included as part of a hub, where a hub is one or more digitally networked Call Processors and PBX systems (as will be discussed later in reference to Figure 8).

The process flow for handling a new telephone call is described below, followed by several examples that illustrate how the CALL PULLBACK mechanism 22 is employed. A caller uses a source telephone 1 to attempt to contact an intended recipient at an office telephone 11. The call originating at the source telephone 1 is switched through the PSTN 3 and routed to the PBX 9 at the office facility 7. The PBX 9 then presents the caller with an inquiry, asking the caller to identify an extension for the office telephone 11. In response, the caller enters an extension and the PBX 9 attempts to route the telephone call to the office telephone 11. If all the lines to the PBX 9 are busy, or if there is a ring, but no answer at the destination telephone 11, or all calls are directly forwarded to the Virtual Network Call Processor 20 using the call forward mechanism in the local telephone equipment, the Virtual Network Call Processor 20 receives the call and subsequently processes the call. Alternatively, the call may be transferred directly into the Virtual Network Call Processor 20 by an operator or other company personnel at the office facility 7. As a further alternative, a caller may dial directly into the

Virtual Network Call Processor 20, or be forwarded in by call forwarding previously set up at the customers Central Office 30C under the following conditions:

<u>Ring no answer on the companies main number or numbers.</u>

An example of this usage could be that no one is available to answer, i.e. after hours, weekends, holidays or the company has suffered a catastrophe in which the facility has been destroyed. A Virtual Voice Network utilizing CALL PULLBACK technology may be programmed to allow designated personnel at a company to call into a Node, enter a password and with a few keystrokes have callers processed to telephones other than those at the company such as the home telephones of company personnel. Even if the company is physically gone, business may still be conducted.

<u>Busy on the company's main number or numbers.</u>

All trunks or lines are busy due to traffic or being busied out at the central office while repair or reprogramming work is being performed.

<u>All calls forward with or without ring reminder.</u>

Some companies provide an after hours courtesy to their callers by taking the time to program their main number so that it forwards to a Virtual Voice Network Node without the caller having to listen to a number of rings.

<u>Forwarding with multiple talk paths.</u>

In the case of a customer location with only one trunk or line, one or more callers may reach a Node at the same time when the given line has multiple talk paths.

When the call is transferred to the Virtual Network Call Processor 20, the Virtual

Network Call Processor 20 recognizes the telephone number that the source telephone 1 was attempting to contact by using direct inward dial, (D.I.D.), automatic number identification, (ANI), or direct number identification system, (DNIS). If the telephone number is associated with an office location, the caller is presented with an options menu (described in audible format) asking the caller to select the person or department with whom the caller wishes to speak. Once selected, the caller is placed on soft hold, while the Virtual Network Call Processor 20 dials an external telephone number and initiates a call progress tone detection operation as will be discussed with respect to Figure 5. The CALL PULLBACK mechanism 22 may then consult a list of stored candidate numbers at which the intended recipient may be located, where the numbers stored are provided by the intended recipient when the intended recipient enters (or updates) a user profile, perhaps when the intended recipient originally subscribes for service. Sequentially, the CALL PULLBACK mechanism 22 attempts to contact the intended recipient at the respective destinations (for example home office 28 or mobile phone 26). If the intended recipient is not located at the first candidate location, the call is "pulled back" and if desired by the customer the CALL PULLBACK mechanism 22 informs the caller that it is about to dial the next location as well as offering the caller other options such as leaving a message. If the caller does nothing, the CALL PULLBACK mechanism 22 may consult from memory the next candidate number, and then attempt to contact the intended recipient at that next candidate number. The CALL PULLBACK mechanism 22 may operate in this fashion until all of the candidate locations have been investigated. If the intended recipient has still not been located, the Virtual Network Call Processor 20 allows the caller the options of leaving a message, contacting an operator, or dialing another extension, for example. On the other hand, if the caller is available at one of the candidate locations, the caller remains on soft hold, while the virtual call network processor 20 presents the intended recipient with a call announcement, such as "this is a call for XYZ Engineering Company, press # to accept or * to reject". If the call is accepted,

the calling party and the intended recipient are connected. If the call is rejected, the calling party is informed that the "name" does not answer and is offered further options, such as speaking with an operator, leaving a voice mail message or dialing another extension for example.

While the calling party is placed on soft hold, the CALL PULLBACK mechanism 22, begins a call process tone detection operation, while dialing the external telephone number, as will be discussed with respect to Figure 5.

The connection that is made by the Virtual Network Call Processor 20 may be made to any phone or device that can be dialed directly, even if the call is placed over the Internet, voice over Internet Protocol. Examples of such devices include cell phones (terrestrial and satellite based), direct inward dial (D.I.D.) telephone numbers, business or home telephone numbers, "Multiserve" or similar service telephone numbers, facsimile devices, computers, etc. A feature of the Virtual Network Call Processor 20 is that the transfer of the call from the Virtual Network Call Processor 20 is made to the intended recipient even though the intended recipient is located on a different PBX than the transferring party.

However, the vast majority of the calls processed are to company departments or fixed locations rather then people who are moving from location to location. In such cases, a call is processed to a given telephone and if not answered, the caller is offered the options of leaving a voice mail message, dialing another extension, dialing 0 for the operator or returning to a portion of the menu where another selection can be made.

Figure 5 is a more detailed block diagram of the CALL PULLBACK mechanism 22 shown in Figure 4. The call pullback mechanism 22 includes an application delay adjustment mechanism 24 as shown in Figure 4, the components of which include a random access memory (RAM) 241, read only memory (ROM) 243, hard disk drive (not shown) and internal interface 245 that are connected to a bus 503. An external interface circuit 501 provides the physical interface, and lower level protocol operations for communicating data between the bus 503 and

the private or public lines 18 and 21, which ultimately connect to the source telephone 1 and PSTN 3 as shown in Figure 4. Additional lines may connect to the external interface 501. A processor 505 is a single processor, although multi-processor architectures, as well as hybrid processor and digital signal processor components may be used as well. Additional processors may be included in the CALL PULLBACK mechanism 22, such as in the application delay adjustment mechanism 24, tone detector 507 and frequency and cadence analyzer 509. Operationally, the transferred call comes in through line 21 to the external interface 501, but alternatively, in a direct dial context, the call may come in directly through line 18. Subsequently, the external interface 501 identifies the party to be called, and retrieves a data file associated with the intended recipient, as identified in the call, by way of the bus 503. Part of data file is a first candidate number, which the CALL PULLBACK mechanism 22 will attempt to contact the intended recipient. The intended recipient's call is placed on hold, while the processor 505 initiates another call on an external line (one of the other lines 18-21).

The tone detector 507 is placed on this external line, so as to determine the call progress status of the call made on the external line. The CALL PULLBACK mechanism 22 places the tone detector 507 on the external line so as to determine if a type of busy signal is present. If the busy signal is present, the calling party is removed from hold and the intended recipient's greeting or, alternatively, the intended recipient's name and condition is audibilized to the calling party. At this time, additional options may be offered to the calling party. However, if no busy signal is detected, the CALL PULLBACK mechanism 22 sends out a series of ticking sounds so the called party will know that this is a call from their Virtual Voice Network and if they choose to wait they will hear the announcement of the type of call they are receiving, example, (Sales verses Customer service) and press # to accept, or * to reject before or after the announcement is made.

The frequency and cadence analyzer 509 characterizes different types of signals from the

local telephone equipment serving the candidate location at which the intended recipient is attempting to be located. The signals to be detected include slow busy signals, fast busy signals, ringing signals, answered signals, and ring no answer signals so that additional candidate locations may be searched and/or the caller may be informed of the status of locating the intended recipient. The frequency and cadence analyzer 509 includes a predetermined set of operations that are intended to interpret the various busy signals, ringing signals, answered signals and the like produced at the far end central office. All frequency and cadence analysis is done on the fly by the frequency and cadence analyzer 509 which consults the application delay table for acceptable cadence values.

As previously discussed, the frequency and cadence of different signals varies between local telephone companies equipment. Accordingly, the frequency and cadence analyzer 509 communicates over the bus 503 to receive the application delay parameters from the application delay adjustment mechanism 24. These parameters are included in the RAM 241, although may also be included in the ROM 243 (conveniently implemented as an EEPROM) or at a remote memory accessible by the internal interface 245, by way of the bus 503.

Alternatively, the frequency and cadence analyzer 509 includes an acoustical signature mechanism that compares respective acoustical signatures against a saved set of acoustical signatures saved in the ROM 243, so as to determine if the response received from the local telephone equipment is a slow busy, a fast busy, etc. The frequency and cadence analyzer 509, also incorporates pattern recognition software that attempts to compare and identify signals received from local telephone equipment, on an on-the-fly basis. If no busy signal is detected, the frequency and cadence analyzer 509 sends out a series of ticking sounds and when the call is answered, a call announcement operation is conducted. The ticking sounds are sent out to alert the called party that their Virtual Voice Network is calling and not someone else. During or after call announcing the called party may then chose to accept or reject the call or if more then one

part of the Network directs calls to them they may choose to listen to the full call announcement before accepting or rejecting the call.

The processor 505 includes an internal memory for program storage and holding intermediate calculation results. However, ROM 243 also includes a number of software objects that are invoked by the processor 505, when analyzing and assessing the respective attributes of the local telephone equipment.

Figure 6 is a flowchart of a process flow for contacting an intended recipient by way of the Virtual Network Call Processor 20 as it implements the CALL PULLBACK mechanism. The process begins in step S51, where the calling party dials the number for the intended recipient at a particular number. The process then proceeds to step S53, where an inquiry is made regarding whether the intended recipient answers the phone call (perhaps by way of a local PBX, such as PBX 9 in the office facility 7 of Figure 4). If the intended recipient answers the phone call, the process proceeds to step S55 where the phone call is connected to the intended recipient, and subsequently the process proceeds to step S65 where all the call processing is completed. Alternatively, steps S51, S53 and S55 may be performed by dialing directly the office number of the intended recipient, thereby bypassing the virtual call network processor 20. If the intended recipient is unavailable to answer, the local central office invokes a call forwarding operation that forwards the call directly to the Virtual Network Call Processor.

A majority of the time a given caller reaches a Virtual Voice Network is because the caller was calling a company rather then an individual. Although no two Virtual Voice Networks need to be identical, the majority of them greet the caller and instruct the caller to enter an extension number or choice. Some extensions do not process calls, some only process calls to a single number all of the time and some process calls to multiple numbers. CALL PULLBACK is invoked when a caller is offered other options after placing an unsuccessful call to a telephone not residing at the same location and not directly connected to the T3i Virtual Voice Network.

Call announcing and dialing multiple phone numbers are enhancements that the customer may or may not wish to use.

If the response to the inquiry in step S53 is negative, the process proceeds to step S57 where the call is forwarded (transferred) to the Virtual Network Call Processor 20, where the Virtual Network Call Processor 20 attempts to contact the intended recipient at one of the predetermined numbers stored at the Virtual Network Call Processor 20. In step S57 the caller is placed on soft hold, while an attempt is made to contact the intended recipient by way of an external line. The process then flows to step S59, where an inquiry is made regarding whether the intended recipient answers the call from the Virtual Network Call Processor at a first number stored in the Virtual Network Call Processor 20. Step S59 may have to be repeated if the intended recipient is not located at the first number and additional numbers are included in the intended recipient's profile that may be automatically checked by the Virtual Network Call Processor 20.

If the response to the inquiry in step S59 is affirmative, the Virtual Network Call Processor announces the call to the intended recipient in step S61. By announcing the call, the intended recipient has the option to receive the telephone call from the calling party, or have the Virtual Network Call Processor inform the calling party that the intended recipient is unable to receive the call. By announcing the call to the intended recipient, the intended recipient knows how to answer the phone, for example, when the calling party's call is taken off soft hold, and connected to the intended recipient's telephone. After step S61, the process proceeds to step S65, where call processing is completed and subsequently the process ends. On the other hand, if the response to the inquiry in step S59 is negative, the Virtual Network Call Processor invokes the CALL PULLBACK mechanism where the call remains on soft hold, and an external line (either the same external line as before, or another line) is used to attempt to contact the intended recipient at the next number identified in the profile of the intended recipient. Using this

example, the CALL PULLBACK mechanism would remove the caller from soft hold and announce that the called party was unavailable and offer further options, one of which could be to dial another number. This process of tracking-down the intended recipient proceeds until all of the candidate locations have been exhausted, at which time the Virtual Network Call Processor either takes a voice mail message, asks the calling party if they would like to identify another person to whom to route the call, etc. Of course, if the CALL PULLBACK mechanism successfully contacts the intended recipient, and the intended recipient decides to receive the call, the caller is then taken off hold, and connected to the intended recipient. Subsequently, the process is completed in step S65 and the process ends.

Application delays are timing values set in the Call Processor portion of the Node. Delays described in the Appendix are used to detect critical tone cadences that the Central Office provides to the Node equipment. These cadences indicate specific call conditions such as a ringback tone indicates that a called number is ringing, and a busy indicates that the called party is busy.

Cadence values are normally set by selecting a PBX type and making modifications to the equipment as needed. As it is not known what PBX type a given Central Office would have or what effect the state of repair or software level would have on the cadences provided, a starting point is to choose the PBX closest to the one which was part of a first node implemented by a user of the system.

After adjusting the Call Processor's cadence recognition to the first Central Office PBX, new adjustments can be made as Central Offices are added and tested to make sure that the new adjustments work with previous Central Office PBXs'. Fail-safe mechanisms are included as features of the CALL PULLBACK to catch any caller that hit an unexpected cadence. These fail-safe mechanisms allow callers the options of reaching a live operator or leaving a message as well as providing first hand intelligence regarding what happened and where an unexpected event

occurred. As the system matures, the fail-safes are not needed as frequently because the system's attributes will become more completely characterized with time.

Problems chiefly occur in areas where the Call Processor detects an answered condition while monitoring a single interrupted-ringback and with slow-busy and fast-busy cadences. When a call is screened, the equipment looks for acceptable cadences for a single interrupted ringback, slow-busy, fast-busy or that the call has been answered. To process a transfer application-delay, indexes are referenced that show the maximum and minimum ON/OFF periods for any tone. If the tone cadence detected does not comply with the ranges set for single ring back, slow busy or fast busy, the Call Processor determines that the call has been answered and the call transfer is completed.

In the case of dead air, such as the Central Office dropping the call, the caller would be removed from soft hold and a fail-safe mechanism would take over. One problem that occurs is when a Central Office recording is played such as an all circuits are busy or that the person being called is out of the area or unavailable. The Call Processor detects an answer and sound, such as someone speaking, and completes the transfer. These occurrences can be kept to a minimum and caller frustration reduced by the following methods:

Prior testing and identification - this allows the network designer to inform the customer of a potential problem and make needed changes or record a special greeting so that the caller has the opportunity to record a message or go to an operator before the called party's number is dialed;

Pulling the call back before the far end recording is played; and

Most customers and callers are used to the recordings being played and are not troubled by them. If the tone cadences are within acceptable ranges, call screening by the called party may be employed and the call accepted or rejected.

Figure 7A is a flowchart of a process for identifying local telephone equipment attributes,

such as frequency and cadence. The process begins in step S661 where the called party is dialed by a node. Subsequently, the process proceeds to step S663, where the cadence and frequency information from signals produced by the local telephone equipment is observed by the node. When observing the cadence and frequency information, the cadence and frequency information is characterized for subsequent processing. The process then proceeds to step S665, where application delays are identified that correspond with the frequency and cadence information that was characterized in step S663. The process then proceeds to step S667, where the node takes appropriate action for the call based on predefined custom parameters and/or reacts to the cadence events, where the reaction is a function of detecting which signals are in fact produced by the local telephone equipment. Subsequently, the process ends.

Figure 7B is a flow chart of a process used by a NODE-OVERTURE Call Processor to screen calls. The process begins in S71, where the call is transferred and the NODE-OVERTURE Call Processor dials the called number and begins looking at tone patterns received from the local telephone equipment. Subsequently, the process proceeds to step S72, where an inquiry is made regarding whether the tones that are received comply with the ranges set by delays 49, 50, 51 or 52, as identified in the appendix attached hereto. If the response to the inquiry in step S72 is negative, the process proceeds to step S73, where the call is considered to have been answered, and subsequently the process ends. However, if the response to the inquiry in step S72 is affirmative, the process proceeds to a ring back inquiry in step S74, where an inquiry is made regarding whether the received tones comply with the ranges set by delays 53, 54, 55 or 56. If the response to the inquiry in step S74 is affirmative, the phone rings and a ring back is monitored. The ring back minimum and maximum tone off period are included in the appendix. The process then concludes.

However, if the response to the inquiry in step S74 is negative, the process proceeds to step S76, where a slow busy inquiry is made. The inquiry in step S76 inquires whether the tones

comply with the ranges set by delays 69, 70, 71 or 72. If the response to the inquiry in step S76 is affirmative, the process proceeds to step S77, where the call is pulled back and the NODE OVERTURE Call Processor speaks the name and condition or greeting and subsequently the process ends. However, if the response to the inquiry in step S76 is negative, the process proceeds to the fast busy inquiry in step S78, and an inquiry is made regarding whether the tones comply with the ranges set by delays 73, 74, 75 or 76. If the response to the inquiry in step S78 is affirmative, the process proceeds to step S79, where the call is pulled back, and an indication is spoken indicating that the call is "invalid" and then the process ends. However, if the response to the inquiry in step S78 is negative, the call is answered in step S80 and then the process ends.

Figure 7C is an exemplary display of information that would be displayed on a monitor screen for a situation where a ring-no-answer operation fails. Application delays that produce a failure are first tested by assigning a mailbox that dials the problem telephone number to a special area code specific object. The port specific print tone trace is activated and the node's Call Processor is called through the port in question. The port will have a speakerphone butt set placed on it so that there is an audible awareness on the part of the person performing the task so as to determine what events are in fact occurring. When answered, the test mailbox is dialed and the tone on and tone off events are monitored in milliseconds as shown on a computer screen. In Figure 7C, a display 701, which, may either be a simultaneous display, or printout or otherwise of a stream of code, illustrates tone information that is displayed for a ring-no-answer operation that fails. As shown in annotation 702, the caller enters DTMF digits, which are the digits associated with the phone to be contacted. In annotation 703, the NODE OVERTURE monitors the line for a tone associated with a dial tone and then subsequently detects the presence of the dial tone. Once the dial tone is detected, and annotation 704 shows that the NODE OVERTURE dials the number associated with the DTMF digits. Annotation 705 indicates that the NODE OVERTURE ignores the first change in tone for a predetermined period of time. Subsequently,

as indicated in annotation 706, the NODE OVERTURE monitors call progress tones from the PBX so as to determine the status of the called extension. Finally, in annotation 707 a failure is indicated when the NODE OVERTURE detects an answer condition because one of the tones of the PBX does not conform to the delays in the application delay table, listed in the appendix.

The tone values in the print tone trace of Figure 7C may be modified to the correct values by using the appropriate commands. For example, in the case of Figure 7C, the application-delay indexes that refer to the error received are indexes 50 and 54, included in the appendix. Note that the failure occurred when the PBX sent a TONE ON for 790 ms. The NODE OVERTURE was set to expect a TONE ON (ring back) for no less than 800 ms and no greater than 1200 ms. This range between 800 ms and 1200 ms is referred to as the "window". In Figure 7C, the window for the silence period (TONE OFF) between adjacent rings is set to no less than 2800 ms and no greater than 3400 ms. The TONE OFF values are within that window.

Figure 8 is a block diagram of an intelligent network of Virtual Network Call Processors (20A, 20B, 20C, 20D) connected together, as shown, in a ring configuration, although other configurations may be performed as well, such as a star or non-geometric specific interconnected configuration. Each of the Virtual Network Call Processors 20A through 20D is configured as several interconnected nodes, a node being a PBX having a Call Processor. The Virtual Network Call Processors 20A-20D are the respective hubs, and thus each serves a different geographical area. As a consequence, a source communication device 1A may connect to the Virtual Network Call Processor 20A (or other Virtual Network Call Processor 20B-20D) by way of the local telephone equipment 30A, or directly to the Virtual Network Call Processor 20A. The Virtual Network Call Processor 20A may then route the information (voice, or other type of data, such as image data, facsimile data, etc.) via the other Virtual Network Call Processors (20B-20D) and then to a destination facility 7 either directly from Virtual Network Call Processor 20D, or via the local telephone equipment 30B.

Because the respective hubs 20A-20D are digitally linked, via dedicated point to point connections or by the use of VPNs' (Virtual Private Networks) or through a gateway server over the Internet, no charges for long distance services are required, although the network of Virtual Network Call Processors 700 certainly could charge a fee for such services or other fee-based links may be used as well.

An example application of the network architecture of Figure 8, might be if either an e-mail message, a facsimile message or other message such as a digitized voice, video or data file were intended to be left with a person in Florida (serviced at Virtual Network Call Processor 20A), a copy of that message may be routed through the network of hubs 20A-20D and to the destination facility 7. Since the connections are by way of dedicated links or VPNs, there are no long distance charges. In addition to the data relaying service, each of the respective Virtual Network Call Processors 20A-20D may also provide the standard call processing features described in Figures 4-5, for example.

The inventive system may include a CALL PULLBACK mechanism that employs a primary rate interface (PRI) that is compatible with National ISDN standards deployment of Simplified Message Desk Interface (SMDI).

The mechanisms and processes set forth in the present description may be implemented using one or more general purpose microprocessors programmed according to the teachings of the present specification, as will be appreciated to those skilled in the relevant art(s). Appropriate software coding can readily be prepared by skilled programmers based on the teachings of the present disclosure, as will be apparent to those skilled in the relevant art(s). The present invention thus also includes a computer-based product which may be hosted on a storage medium and include instructions that can be used to program a computer to perform a process in accordance with the present invention. The storage medium can include, but is not limited to, any type of disk including floppy disk, optical disk, CDROM, magneto-optical disk, ROMs,

RAMs, EPROMs, EEPROMs, flash memory, magnetic or optical cards, or any type of media suitable for storing electronic instructions, either locally or remotely.

Obviously, numerous modifications and variations of the present invention are possible in light of the above teachings. It is therefore to be understood that within the scope of the appended claims, the invention may be practiced otherwise than as specifically described herein.

CLAIMS:

1. A virtual network call processing system, comprising:

a communication line interface configured to be connected to a source terminal and receive a calling message from the source terminal directed to an intended recipient;

a call processor with a call pullback mechanism including,

a data processor, and

a computer readable memory having computer readable instructions encoded therein that when executed by said data processor implement a local telephone equipment characterization mechanism that characterizes signaling attributes of signals produced by local telephone equipment that service different geographical locations at which the intended recipient may be located; and

a signal determination mechanism configured to determine whether the signals provided by the local telephone equipment have at least one of a frequency and cadence associated with a signal event that includes at least one of a fast busy signal, slow busy signal, ringing signal, answered signal, and ring-no-answer signal.

2. The system according to Claim 1, further comprising:

an error handling mechanism configured to process the calling message when the signal determination mechanism fails to determine that the signal event occurred.

3. The system according to Claim 1, wherein:

said signal determination mechanism includes a software tool programmed to recognize the at least one of the frequency and cadence associated with the signal event from signals associated with the local telephone equipment.

4. The system of Claim 1, wherein:

said computer readable memory includes

an intended recipient profile, having a first destination number and a second destination number, and

said call pullback mechanism further includes a recipient contact mechanism being configured to attempt to first contact said intended recipient via an external line at the first destination number, and if not present, being configured to attempt to contact said intended recipient at the second destination number.

5. The system of Claim 4, wherein:

said signal determination mechanism includes a tone detector being configured to detect when said calling message on said communication line is answered, or said signal event occurs; and

said system further includes a call announcing mechanism configured to execute a whisper transfer to the called party that allows the called party to accept or reject the calling message.

6. The system of Claim 5, wherein:

said tone detector being configured to notify the recipient contact mechanism when said communication line is not answered so said recipient contact mechanism proceeds to contact said intended recipient at said second destination number.

7. The system of Claim 1, wherein:

said local telephone equipment characterization mechanism includes a frequency characterization mechanism, configured to characterize a frequency of the signals produced by

respective of said local telephone equipment.

8. The system of Claim 1, wherein:

said local telephone equipment characterization mechanism includes a cadence analyzer configured to analyze a cadence of respective of the signals provided by respective of said local telephone equipment.

9. The system of Claim 8, wherein:

said local telephone equipment characterization mechanism includes a frequency analyzer configured to analyze a frequency of the signals provided by respective of said local telephone equipment.

10. The system of Claim 1, further comprising:

a signal feature normalization mechanism, including an application delay adjustment mechanism configured to adjust respective application delays in said call processor so as to standardize signal attributes provided by respective of the local telephone equipment.

11. The system of Claim 10, further comprising:

another data processor and another computer readable memory configured to implement another local telephone equipment characterization mechanism and another signal feature normalization mechanism, said processor and computer readable medium being connected to said another processor and said another computer readable medium by an intercity communication link.

12. A method for processing a call in a virtual network call processing system,

comprising the steps of:

receiving a calling message from a source terminal directed to an intended recipient;

retrieving a data profile of the intended recipient from a computer readable medium;

identifying a number to contact the intended recipient via a local telephone equipment;

characterizing signal attributes of signals provided by the local telephone equipment;

initiating the call on an external line with said number at said local telephone equipment;

normalizing the signal from the local telephone equipment;

transferring the calling message if the call is accepted by the intended recipient, but retaining the calling message for future processing if the call is not accepted by the intended recipient.

13. The method of Claim 12, further comprising the steps of:

identifying another number in said data profile of said intended recipient;

calling on at least one of the external line and another external line, said another number at another local telephone equipment;

normalizing a signal from the another local telephone equipment;

transferring the calling message if accepted by the intended recipient at the another number, but retaining the calling message for further processing if not accepted.

14. The method of Claim 13, further comprising the step of:

producing a tone on said external line and detecting when said external line is answered.

15. The method of Claim 12, wherein:

said characterizing step comprises characterizing a frequency of the signal provided by the local telephone equipment.

16. The method of Claim 12, wherein:

said characterizing step comprises characterizing a cadence of the signal provided by the local telephone equipment.

17. The method of Claim 16, wherein:

said characterizing step further comprises characterizing a frequency of the signal provided by the local telephone equipment.

18. The method of Claim 12, wherein:

said normalizing step comprises adjusting respective application delays so as to standardize signal attributes of the signal from the local telephone equipment.

19. The method of Claim 12, wherein:

said calling step comprises passing the calling message from a first hub to a second hub, prior to reaching the local telephone equipment.

20. A computer readable medium encoded with computer readable instructions for use in a system having a communication line interface configured to be connected to a source terminal and configured to receive a calling message from the source terminal directed to an intended recipient, said computer readable instructions when executed by a data processor implement a system comprising:

a local telephone equipment characterization mechanism that characterizes signaling attributes of signals produced by local telephone equipment that service different geographical locations at which the intended recipient is located;

a signal determination mechanism configured to determine whether the signals provided

by the local telephone equipment have at least one of a frequency and cadence associated with a signal event that includes at least one of a fast busy signal, slow busy signal, ringing signal, answered signal, and ring-no-answer signal; and

a call pullback mechanism configured to call an intended recipient and transfer a calling message to said intended recipient if said signal determination mechanism determines said call is answered, but not transferring said calling message if said signal determination mechanism determines that said call is not answered.

21. A virtual network call processing system, comprising:

means for receiving a calling message from a source terminal directed to an intended recipient;

means for identifying a number to contact the intended recipient via a local telephone equipment;

means for characterizing signal attributes of signals provided by the local telephone equipment;

means for initiating a call on an external line with said number at said local telephone equipment;

means for normalizing the signal from the local telephone equipment; and

means for transferring the calling message if the call is accepted by the intended recipient, but retaining the calling message for future processing if the call is not accepted by the intended recipient.

CALL PROCESSING SYSTEM, METHOD AND
COMPUTER PROGRAM PRODUCT

ABSTRACT OF THE DISCLOSURE

A system, method and computer program product implement a Virtual Network Call Processor with a CALL PULLBACK mechanism for providing a type of screened call transfer. Callers, while attempting to contact an intended recipient, have their calls sent to the Virtual Network Call Processor, which places the caller on soft hold while attempting to locate the intended recipient. The Call Processor uses another external line to call the intended recipient at one of a number of predetermined locations identified by stored numbers where each number is serviced by perhaps different local telephone equipment having different characteristics and attributes. The CALL PULLBACK mechanism is used to identify signaling attributes of signals provided by the respective local telephone equipment, by analyzing frequency and cadence information from the signals and normalize the signals so as to detect a status of the Call Processor's attempt to reach the intended recipient. The signaling attributes and customer-specific information are controlled by objects, which are well thought out preprogrammed and proven software constructs that simplify programming and ensure reliable operations. The calling party is kept on soft hold while the intended recipient of the call is attempted to be contacted at the different locations. If the CALL PULLBACK mechanism determines that the signals provided by the local telephone equipment, after being normalized, indicate the intended recipient does not pick up the call, the CALL PULLBACK mechanism attempts to reach the intended recipient at another one of the numbers, all the while the calling party is kept on soft hold. In this way, the global Virtual Network Call Processor, is capable of servicing not only individuals and companies serviced by a single PBX with a call process, but also for any number of other users not serviced by the PBX.

APPENDIX

APPLICATION DELAY TABLE.INDEX DELAY (msec.)

0	0
1	7000
2	5000
3	500
4	5000
5	1200
6	1000
7	30000
8	800
9	4000
10	6000
11	1000
12	1000
13	10000
14	20000
15	25000
16	2000
17	20
18	1000
19	200
20	240
21	100
22	140
23	200
24	260
25	300
26	400
27	260
28	460
29	800
30	900
31	1200
32	1500

33	2000	
34	2100	
35	2700	
36	3900	
37	4000	
38	6000	
39	1000	
40	4000	
41	10000	
42	1500	
43	1500	
44	2000	
45	500	
46	1700	
47	700	
48	500	
49	2500	Maximum tone on period for any tone. When ringback, busy or fast busy is encountered this delay is used to determine whether the tone on is a valid tone. When an encountered tone on is longer then this delay it is assumed to be an answer.
50	50	
51	4600	Maximum tone off period for any tone. When ringback, busy or fast busy is encountered this delay is used to determine whether the tone off is of valid duration. When tone off is greater then this value it is assumed to be an answer.
52	40	
53	2500	Maximum tone on period. This is the longest tone on event that will be considered as ringback. This application delay is used to determine whether the tone on cadence event being monitored is ringback. If the tone on event is longer then this application delay it is assumed to not be ringback.

54	800	
55	4600	Ringback maximum tone off period. This is the longest tone off event that will be considered as ringback. If the tone off event being monitored is shorter then this application delay it is assumed to not be ringback.
56	2000	Ringback minimum tone off period. This is the shortest tone off event to qualify as ringback. This application delay is used to determine whether the tone cadence being monitored is ringback. If the tone cadence being monitored is less in duration then this value it is assumed to not be ringback
57	2300	
58	20	
59	15000	
60	20	
61	500	
62	300	
63	500	
64	300	
65	3200	
66	2800	
67	280	
68	120	
69	600	
70	400	
71	600	
72	400	
73	320	
74	180	
75	320	
76	180	

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TITLE OF THE INVENTION

CALL PROCESSING SYSTEM, METHOD AND COMPUTER PROGRAM PRODUCT

CROSS REFERENCE TO RELATED APPLICATIONS

The present document claims the benefit of the earlier filing date of, and contains subject matter related to that disclosed in, co-pending U.S. provisional application Serial No. 60/082,730 filed April 23, 1998, having common inventorship, the entire contents of which being incorporated herein by reference.

BACKGROUND OF THE INVENTION

Field of the Invention:

The present invention pertains to call processing systems, methods and computer-based products used for telephony systems in which calls may be screened by a called party prior to connection. More particularly, the present invention is directed to voice and data systems that include Call Processors and Gateway Servers among other data communication resources, in which calls targeted for a predetermined location are directed by the programming of a Virtual Voice Network. This direction can be to any device that can be directly dialed, such as a telephone including cell phone, fax machine or modem even if the call for the targeted device is processed across the Internet, (voice or fax over Internet Protocol.)

Discussion of the Background

Advances in modern electronics and digital communication enable individuals to communicate virtually anywhere around the world. With the advent of cellular telephones,

personal communication services, and satellite telephony, individuals in advanced as well as developing societies have an expectation of being able to communicate with others anytime and anywhere in a smooth and seamless fashion. The bulk of existing communication infrastructure is provided by local, regional and long distance telephone companies, i.e., the public switch telephone network (PSTN), which uses land lines, among other resources, for providing point-to-point communications, where each point is identifiable by a separate telephone number. For example, a caller may use a first telephone number when attempting to reach a person at the person's home, but uses a second number for contacting the person at the person's office.

A recent challenge has been how to use the PSTN, components of which contain old technology, to provide the flexibility to support people who want to remain accessible while being mobile. In light of this backdrop, some corporations use private branch exchanges (PBX) at the corporation facilities to provide "smart" functions for handling incoming phone calls (described generically here, but referring to both voice and data calls) for the convenience of its customers. Using these functions, even if an employee is not available, a properly equipped PBX enables an outside caller to be conveniently patched into voice mail, routed to another number, or perhaps transferred to a different facility in an attempt to handle the call in a user-friendly environment.

Figure 1 is a block diagram of a conventional PSTN and PBX based system that enables a source telephone 1 to communicate with a destination telephone 11, of an intended recipient. In Figure 1, the source telephone 1 connects to the PSTN 3 via a line (wired or wireless). The PSTN 3 recognizes the telephone number input at the source telephone 1 and provides the switch infrastructure to ultimately connect the caller with the PBX 9, which has the burden of providing the "operator" interface functions. In many cases, the corporation facilities 7 may incorporate a relay Call Processor with auto attendant functions 13 even though unit costs for such devices could exceed 1.5 million dollars in 1998. An example of such a Call Processor is an

OVERTURE 300 sold by the Lucent/Octel Messaging Division.

The relay Call Processor with auto attendant 13 operates when the call is received by the PBX 9 and attempts to ring the destination telephone 11, while placing the caller on hold or utilizing any of a number of types of integration depending on the makes and models of the equipment. If the destination telephone 11 is not picked-up after a predetermined number of rings, the Call Processor with auto attendant 13 initially reports a message to the caller such as "thank you for calling company A. John Doe is on the telephone so please leave a message, dial another extension, or dial 0 for the operator." The Call Processor with auto attendant 13 is able to handle the call for the employee in this way because the PBX 9 receives and routes all telephone calls within the company facility 7 without having to interface with a variety of different local telephone equipment, each having unique signaling attributes.

For users that do not have the benefit of a corporation's PBX 9, the PSTN offers users a call forwarding operator 5 that, at the instruction of the intended recipient, forwards incoming calls to a secondary number when the intended recipient is unavailable at a primary number. This call forwarding mechanism however employs equipment at the PSTN and does not offer the same degree of convenient voice mail and auto attendant functions offered by the PBX 9 at the company facility 7.

As presently recognized by the inventor, the PBX 9 is an inherently "local" device hosted at a certain destination facility, such as a company. Available for equipment of such expense, smaller devices such as the relay Call Processor with auto attendant 13 are included with the PBX 9 to provide added functionality. Adding to the expense, the relay Call Processor with auto attendant 13 must be customized by technicians when installed at the company facility 7 so as to be compatible with the local telephone company equipment if any screened transfer types of calls were to be placed to external telephone numbers. Customization is needed because the PSTN 3 is not homogenous, but rather made up of numerous equipment of local telephone companies that

may or may not have the same equipment. As an example of different signaling attributes of signals provided by typical telephone equipment, the frequency and cadence of slow-busy signals (or other signals, as will be discussed) may be substantially different from one local telephone to the next. Similarly, other signals such as a fast busy signal, indicating an error is present, differs as well.

Figure 2 is a timing diagram of a ring/silence signal offered by exemplary local telephone company equipment. A high voltage level indicates a ring interval, while a lower voltage indicates a silence interval. For illustrative purposes, the interval "A" may typically range between a maximum tone-on (i.e., ring interval) of 1,200 ms to a minimum of 800 ms, while a typical number may be 1000 ms. The interval "B"(a silence interval) may range between 3500 ms and 2801 ms, with a typical number being 2881 ms. Interval "C" may typically range between 1200 ms and 800 ms, with a typical number being 942 ms. Interval "D" may typically range between 3485 ms and 2899 ms, with a typical number being 2910 ms. Similarly, the interval "E" may typically range between 1200 and 800 ms with a typical time 785 ms (which is less than the stated lower end of the "typical" range, but included to show that it is nonetheless a possibility). Due to this variation in cadence and frequency between signals provided by local telephone equipment, generic relay Call Processors with auto attendant functions are conventionally believed to require the use of technicians to personally customize the "application delays". This approach essentially normalizes the cadence and frequency terms so that the Call Processor can effectively interface with that particular local telephone equipment. Consequently, according to conventional wisdom, it is not believed wise, nor even possible, to use a relay Call Processor with auto attendant function in a central location that operates with different local telephone equipment because the diversity of telephone equipment does not permit the relay Call Processor with auto attendant to handle common signals in a like fashion.

Figure 3 is a flowchart of an example method of how a caller at a source telephone 1 (Fig.

1) attempts to communicate with an intended recipient at the company facility 7. The process begins in step S1, where the caller initiates a call to the intended recipient by dialing a phone number of the company where the intended recipient is believed to be located. The process then proceeds to step S3, where the call is answered by the PBX 9 at the company facility 7, and the PBX 9 passes the call to the Call Processor with auto attendant 13. The process then proceeds to step S5, where the caller is requested to dial the extension of the intended recipient. The process then proceeds to step S7 where an inquiry is made regarding whether the individual identified at that extension is available. While making the inquiry, the Call Processor in the PBX 9 places the caller on hold and rings the destination telephone a predetermined number of times. If the intended recipient does not answer the telephone call after the predetermined number of times or if a busy signal is received, the Call Processor concludes that the intended recipient is unavailable. If the response to the inquiry in step S7 is affirmative, the PBX 9 connects the caller with the intended recipient in step S9 and the process then proceeds to step S11 where the call is completed and then the communication session ends. However, if the response to the inquiry in step S7 is negative, the process proceeds to step S13 where the relay Call Processor with auto attendant 13, audibly presents a set of options to the caller. Typical options include leaving a voice mail message, hitting zero to dial an operator or entering the extension of another party. Once the options are presented, the process proceeds to S15 where the caller selects an option and then in step S17 the selected option is executed. Subsequently the process ends. The Call Processor may also offer other options to the caller, such as attempting to contact the intended recipient at another location. If the caller chooses this option, the Call Processor with auto attendant 13 performs a blind transfer to that other location. Since the Call Processor with auto attendant 13 performs the blind transfer, the Call Processor performs no additional processing of the call even if the intended recipient is not available at the other location. The blind transfer will be made regardless; even if the called party is busy, ring no answer, error tone or dead air.

Some coverage methods employ various call forwarding schemes in the event a called device is busy, or ring no answer. These methods are designed to forward the caller to a receptor mailbox. Often the receptor mailbox is located where the call originated and the called party pays the bill for any forwarding or long distance charges.

As identified by the present inventor, a limitation with conventional devices and methods is that the functions offered by the Call Processor in the PBX 9 are prohibitively expensive for the "small user". In other words, the call processing functions available at the company facility 7 are expensive to purchase and install, and thus are unsuitable for private use. Furthermore, due to differences between the different types of local telephone company equipment employed throughout the PSTN, making a conventional relay Call Processor with auto attendant available to users across a number of different local telephone company equipment is not believed to be possible, due to the different signaling attributes of the equipment employed by the different local telephone companies.

As presently recognized, the installation procedures of PBX 9 with the relay Call Processor are complex in that "hands on" customization and testing of the local telephone equipment is believed to be required in conventional systems when adjusting the destination delays for the relay Call Processor. Such difficulties are factors that contribute to the expense of purchasing and maintaining a Call Processor, even though conventional Call Processors are used over a specific geographical region sharing a common set of telephone equipment.

More centralized functions, such as call forwarding operations provided by the PSTN are incapable of detecting whether a person is available at one of the candidate locations, and "pulling back" the call for further processing if the person is unavailable. Moreover, the call forwarding operations perform a blind transfer of the call, and do not wait to determine whether the destination party will in fact receive the call. As a consequence, the user-friendliness of the call forwarding operation is presently viewed as being sub-optimal.

U.S. Patent No. 5,375,161 describes a telephone control system with branch routing, which includes a call conferencing feature (see, e.g., Figure 14', step 1419) that waits to determine whether or not a user may be located at another number. This technique thus employs precision busy/ring detection that requires *a priori* knowledge of the attributes of the local telephone communication equipment. Without this knowledge, it would not be possible for such a device to operate without significant customization of PSTN equipment at varying locations. Furthermore, the precision busy/ring would not be able to recognize an error tone, which has the same frequency as a busy but a different cadence, because the precision busy/ring unit monitors only frequency, not cadence. Cadence is a variable that fluctuates most from Central Office to Central Office. When considering a conference feature with voice, a number of conditions should be taken into consideration, including hardware sensitivity and ability to be configured, susceptibility to talk off in which the human voice emulates touch tone, and background noise. Regarding background noise, a PC modem, for example, connecting to a service provider for Internet access can cause any touch-tone activated equipment to do unexpected things. Thus, conferencing features are suboptimal.

SUMMARY OF THE INVENTION

Accordingly, a feature of the present invention is to provide a novel system, method and computer based product that overcomes the limitations of the conventional methods and systems discussed above. While a full description of the invention and its various features are described in the following section, a brief, non-exhaustive description of features of the present invention is now described. A facet of this invention is that the Call Processors used normally do not reside at the same location and are not directly connected to a customer's PBX or to the customer's Central Office. If the called party is busy / no answer or an error tone is encountered, the calling party is informed of the status of the called party and may be offered further options.

A "CALL PULLBACK" mechanism is included in a central location (i.e., accessible to geographically separated users) in a Call Processor, which is a component of a virtual Call Processor network. The Call Processor in the virtual Call Processor network places a caller on "soft hold" while attempting to contact the intended recipient at one of various predetermined numbers. In order to overcome the incompatibility issue of operating with different local telephone equipment, a feature of the present invention is a frequency and cadence detection mechanism that is able to detect different characteristics of slow busy, fast busy, ringing, answered, and ring no answer tones as provided by different local telephone equipment. To this end, the Call Processor of the present invention associates different frequencies and cadences with various events occurring with candidate numbers at which the intended recipient may be located. In the case of a call placed through a Gateway Server across the Internet the frequency and cadence detection may be performed by equipment located at the far end point of presence (POP) with that equipment notifying the originating Call Processor of the status of the call. Accordingly, a feature of the present invention is the establishment of acceptable ranges of frequency and cadence attributes of signals from various local telephone company equipment that service the respective candidate telephone numbers. To this end, the system incorporates a method for implementing the CALL PULLBACK mechanism.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete appreciation of the invention and many of the attendant advantages thereof will be readily obtained as the same becomes better understood by reference to the following detailed description when considered in connection with the accompanying drawings, wherein:

Figure 1 is a system level block diagram of a conventional telephony network that includes a Call Processor at a destination facility;

Figure 2 is a timing diagram illustrating an exemplary variation in cadence and frequency of signals provided by different local telephone equipment;

Figure 3 is a flowchart of a method for handling a call in a conventional relay Call Processor;

Figure 4 is a system level block diagram of a Virtual Network having a central Call Processor according to the present invention;

Figure 5 is a block diagram of components in the CALL PULLBACK mechanism according to the present invention;

Figure 6 is a flowchart of a process for contacting an intended recipient by way of the Virtual Network and implementing the CALL PULLBACK mechanism according to the present invention;

Figure 7A is a flowchart of a process for identifying and associating local telephone equipment attributes with candidate customer numbers stored in a computer readable medium according to the present invention;

Figure 7B is a flowchart of a call screening process employed by a Lucent/Octel Node-Overture Call Processor;

Figure 7C is an annotated tone information screen for a failed ring-no-answer; and

Figure 8 is a block diagram of a network of interconnected Virtual Networks that enable both voice and fax messages and other signals to be transported from a source terminal to a destination facility.

BRIEF DESCRIPTION OF THE APPENDIX

An appendix is attached hereto, that contains an application delay table with an index of available delays.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring now to the drawings, wherein like reference numerals designate identical or corresponding parts throughout the several views, Figure 4 is a system level block diagram of a network 400 according to the present invention. A feature of the network 400 is a Virtual Network Call Processor 20 that is separated from the company owned facility 7 located on property owned by an employee of a customer who subscribes to the network 400. The Virtual Network Call Processor may be implemented as a VIRTUAL VOICE NETWORK NODE, offered by TOUCHTONE TECHNOLOGIES Inc. (T3i) and includes a variety of equipment, including a switch, one or more Call Processors with on-board IVR units, multiple T1-spans and or a Gateway Server or Servers that may reside on a network (such as a local area network, LAN) with other equipment. As will become clear, the Virtual Network Call Processor 20 may also operate completely independently of equipment owned and operated by private corporations, and may be used to provide Call Processor, auto attendant, IVR and facsimile functions for individuals with no access to corporate PBX resources. The Virtual Network Call Processor 20 may also be adapted to provide plug-in applications such as Unified Messaging where e-mail may be stored in a mailbox along with voice and fax messages. These e-mail messages may then be read by the voice server to the subscriber. In particular, the Virtual Network Call Processor 20 connects via private or public lines 18 to a source telephone 1. The private or public lines 18 may be part of the PSTN 3, or private lines owned or leased by individual consumers. While the term "lines" is used, these lines may also be wireless links such as private microwave links, or terrestrial or space-based cellular and wireless communication links, or an Internet Backbone employing Gateway Servers for example. Furthermore, the source telephone 1 need not be a conventional telephone, but may also be other communication devices that transmit data from one location to another such as a facsimile device, computer, computer telephone or Internet accessible terminal, for example.

The Virtual Network Call Processor 20 connects to both the source telephone 1, as well as the PSTN or Internet Backbone through a Gateway Server 3, by way of communication links 21, which may be private or leased lines, for example. The source telephone 1, also connects directly to the PSTN 3, which is made up of an interconnected network of equipment owned by companies that service different regions of the United States (or the equivalent of other national and private communication networks in other countries). The PSTN 3, illustrated in Figure 4, includes an interconnected network of three sets of local telephone equipment 30A, 30B and 30C, located in three geographically distinct regions (region 1 - region 3). As previously discussed, the local telephone equipment 30A-30C are often different systems that have different signaling attributes. For example, the telephone equipment 30A may produce a fast busy signal with different signal features than that of local telephone equipment 30C. In particular, as presently recognized, the difference may be in the form of frequency and cadence differences, where "frequency" refers to signal pitch and "cadence" refers to a rhythm of the respective on and off tone cycles that form a beat. Thus, local telephone equipment 30A, which may service a home office 28 may have distinctive frequency and cadence characteristics as compared with that of the local telephone equipment 30B that services the intended recipient's mobile telephone 26, or the local telephone equipment 30C that services the office telephone 11 at the office facility 7. While the PBX 9 at the office facility 7 can receive the phone call directly from source telephone 1, the PBX 9 is capable of only transferring the call internally with devices connected to the PBX 9 or relying on a call forwarding mechanism 5 offered by the local telephone equipment 30C (see, e.g., Figure 1).

However, by subscribing to services offered by the Virtual Network Call Processor 20, the intended recipient is given the option to invoke the CALL PULLBACK mechanism 22 in the Virtual Network Call Processor 20, which, if desired, allows the user to have calls sent to one of any number of candidate locations, each of which may or may not be serviced by different local

telephone equipment. Moreover, because the Virtual Network Call Processor 20 is centrally located (i.e., accessible to parties external to a company's PBX 9), the Virtual Network Call Processor 20 is available for use by many different users, not just users of the PBX 9. Each user can have phone calls that originate at the source telephone 1 be forwarded to the Virtual Network Call Processor 20 and thereby invoke the CALL PULLBACK mechanism 22. The CALL PULLBACK mechanism enables a screened type of call transfer, as compared to a blind transfer where the call is sent to one of several candidate locations without regard for whether the user actually picks up the transferred call at that location.

The Virtual Network Call Processor 20 is shown as part of a node that is made up of a Call Processor, PBX, IVR and other equipment. However, the node may be included as part of a hub, where a hub is one or more digitally networked Call Processors and PBX systems (as will be discussed later in reference to Figure 8).

The process flow for handling a new telephone call is described below, followed by several examples that illustrate how the CALL PULLBACK mechanism 22 is employed. A caller uses a source telephone 1 to attempt to contact an intended recipient at an office telephone 11. The call originating at the source telephone 1 is switched through the PSTN 3 and routed to the PBX 9 at the office facility 7. The PBX 9 then presents the caller with an inquiry, asking the caller to identify an extension for the office telephone 11. In response, the caller enters an extension and the PBX 9 attempts to route the telephone call to the office telephone 11. If all the lines to the PBX 9 are busy, or if there is a ring, but no answer at the destination telephone 11, or all calls are directly forwarded to the Virtual Network Call Processor 20 using the call forward mechanism in the local telephone equipment, the Virtual Network Call Processor 20 receives the call and subsequently processes the call. Alternatively, the call may be transferred directly into the Virtual Network Call Processor 20 by an operator or other company personnel at the office facility 7. As a further alternative, a caller may dial directly into the

Virtual Network Call Processor 20, or be forwarded in by call forwarding previously set up at the customers Central Office 30C under the following conditions:

Ring no answer on the companies main number or numbers.

An example of this usage could be that no one is available to answer, i.e. after hours, weekends, holidays or the company has suffered a catastrophe in which the facility has been destroyed. A Virtual Voice Network utilizing CALL PULLBACK technology may be programmed to allow designated personnel at a company to call into a Node, enter a password and with a few keystrokes have callers processed to telephones other than those at the company such as the home telephones of company personnel. Even if the company is physically gone, business may still be conducted.

Busy on the company's main number or numbers.

All trunks or lines are busy due to traffic or being busied out at the central office while repair or reprogramming work is being performed.

All calls forward with or without ring reminder.

Some companies provide an after hours courtesy to their callers by taking the time to program their main number so that it forwards to a Virtual Voice Network Node without the caller having to listen to a number of rings.

Forwarding with multiple talk paths.

In the case of a customer location with only one trunk or line, one or more callers may reach a Node at the same time when the given line has multiple talk paths.

When the call is transferred to the Virtual Network Call Processor 20, the Virtual

Network Call Processor 20 recognizes the telephone number that the source telephone 1 was attempting to contact by using direct inward dial, (D.I.D.), automatic number identification, (ANI), or direct number identification system, (DNIS). If the telephone number is associated with an office location, the caller is presented with an options menu (described in audible format) asking the caller to select the person or department with whom the caller wishes to speak. Once selected, the caller is placed on soft hold, while the Virtual Network Call Processor 20 dials an external telephone number and initiates a call progress tone detection operation as will be discussed with respect to Figure 5. The CALL PULLBACK mechanism 22 may then consult a list of stored candidate numbers at which the intended recipient may be located, where the numbers stored are provided by the intended recipient when the intended recipient enters (or updates) a user profile, perhaps when the intended recipient originally subscribes for service. Sequentially, the CALL PULLBACK mechanism 22 attempts to contact the intended recipient at the respective destinations (for example home office 28 or mobile phone 26). If the intended recipient is not located at the first candidate location, the call is "pulled back" and if desired by the customer the CALL PULLBACK mechanism 22 informs the caller that it is about to dial the next location as well as offering the caller other options such as leaving a message. If the caller does nothing, the CALL PULLBACK mechanism 22 may consult from memory the next candidate number, and then attempt to contact the intended recipient at that next candidate number. The CALL PULLBACK mechanism 22 may operate in this fashion until all of the candidate locations have been investigated. If the intended recipient has still not been located, the Virtual Network Call Processor 20 allows the caller the options of leaving a message, contacting an operator, or dialing another extension, for example. On the other hand, if the caller is available at one of the candidate locations, the caller remains on soft hold, while the virtual call network processor 20 presents the intended recipient with a call announcement, such as "this is a call for XYZ Engineering Company, press # to accept or * to reject". If the call is accepted,

the calling party and the intended recipient are connected. If the call is rejected, the calling party is informed that the "name" does not answer and is offered further options, such as speaking with an operator, leaving a voice mail message or dialing another extension for example.

While the calling party is placed on soft hold, the CALL PULLBACK mechanism 22, begins a call process tone detection operation, while dialing the external telephone number, as will be discussed with respect to Figure 5.

The connection that is made by the Virtual Network Call Processor 20 may be made to any phone or device that can be dialed directly, even if the call is placed over the Internet, voice over Internet Protocol. Examples of such devices include cell phones (terrestrial and satellite based), direct inward dial (D.I.D.) telephone numbers, business or home telephone numbers, "Multiserve" or similar service telephone numbers, facsimile devices, computers, etc. A feature of the Virtual Network Call Processor 20 is that the transfer of the call from the Virtual Network Call Processor 20 is made to the intended recipient even though the intended recipient is located on a different PBX than the transferring party.

However, the vast majority of the calls processed are to company departments or fixed locations rather then people who are moving from location to location. In such cases, a call is processed to a given telephone and if not answered, the caller is offered the options of leaving a voice mail message, dialing another extension, dialing 0 for the operator or returning to a portion of the menu where another selection can be made.

Figure 5 is a more detailed block diagram of the CALL PULLBACK mechanism 22 shown in Figure 4. The call pullback mechanism 22 includes an application delay adjustment mechanism 24 as shown in Figure 4, the components of which include a random access memory (RAM) 241, read only memory (ROM) 243, hard disk drive (not shown) and internal interface 245 that are connected to a bus 503. An external interface circuit 501 provides the physical interface, and lower level protocol operations for communicating data between the bus 503 and

the private or public lines 18 and 21, which ultimately connect to the source telephone 1 and PSTN 3 as shown in Figure 4. Additional lines may connect to the external interface 501. A processor 505 is a single processor, although multi-processor architectures, as well as hybrid processor and digital signal processor components may be used as well. Additional processors may be included in the CALL PULLBACK mechanism 22, such as in the application delay adjustment mechanism 24, tone detector 507 and frequency and cadence analyzer 509. Operationally, the transferred call comes in through line 21 to the external interface 501, but alternatively, in a direct dial context, the call may come in directly through line 18. Subsequently, the external interface 501 identifies the party to be called, and retrieves a data file associated with the intended recipient, as identified in the call, by way of the bus 503. Part of data file is a first candidate number, which the CALL PULLBACK mechanism 22 will attempt to contact the intended recipient. The intended recipient's call is placed on hold, while the processor 505 initiates another call on an external line (one of the other lines 18-21).

The tone detector 507 is placed on this external line, so as to determine the call progress status of the call made on the external line. The CALL PULLBACK mechanism 22 places the tone detector 507 on the external line so as to determine if a type of busy signal is present. If the busy signal is present, the calling party is removed from hold and the intended recipient's greeting or, alternatively, the intended recipient's name and condition is audibilized to the calling party. At this time, additional options may be offered to the calling party. However, if no busy signal is detected, the CALL PULLBACK mechanism 22 sends out a series of ticking sounds so the called party will know that this is a call from their Virtual Voice Network and if they choose to wait they will hear the announcement of the type of call they are receiving, example, (Sales verses Customer service) and press # to accept, or * to reject before or after the announcement is made.

The frequency and cadence analyzer 509 characterizes different types of signals from the

local telephone equipment serving the candidate location at which the intended recipient is attempting to be located. The signals to be detected include slow busy signals, fast busy signals, ringing signals, answered signals, and ring no answer signals so that additional candidate locations may be searched and/or the caller may be informed of the status of locating the intended recipient. The frequency and cadence analyzer 509 includes a predetermined set of operations that are intended to interpret the various busy signals, ringing signals, answered signals and the like produced at the far end central office. All frequency and cadence analysis is done on the fly by the frequency and cadence analyzer 509 which consults the application delay table for acceptable cadence values.

As previously discussed, the frequency and cadence of different signals varies between local telephone companies equipment. Accordingly, the frequency and cadence analyzer 509 communicates over the bus 503 to receive the application delay parameters from the application delay adjustment mechanism 24. These parameters are included in the RAM 241, although may also be included in the ROM 243 (conveniently implemented as an EEPROM) or at a remote memory accessible by the internal interface 245, by way of the bus 503.

Alternatively, the frequency and cadence analyzer 509 includes an acoustical signature mechanism that compares respective acoustical signatures against a saved set of acoustical signatures saved in the ROM 243, so as to determine if the response received from the local telephone equipment is a slow busy, a fast busy, etc. The frequency and cadence analyzer 509, also incorporates pattern recognition software that attempts to compare and identify signals received from local telephone equipment, on an on-the-fly basis. If no busy signal is detected, the frequency and cadence analyzer 509 sends out a series of ticking sounds and when the call is answered, a call announcement operation is conducted. The ticking sounds are sent out to alert the called party that their Virtual Voice Network is calling and not someone else. During or after call announcing the called party may then chose to accept or reject the call or if more then one

part of the Network directs calls to them they may choose to listen to the full call announcement before accepting or rejecting the call.

The processor 505 includes an internal memory for program storage and holding intermediate calculation results. However, ROM 243 also includes a number of software objects that are invoked by the processor 505, when analyzing and assessing the respective attributes of the local telephone equipment.

Figure 6 is a flowchart of a process flow for contacting an intended recipient by way of the Virtual Network Call Processor 20 as it implements the CALL PULLBACK mechanism. The process begins in step S51, where the calling party dials the number for the intended recipient at a particular number. The process then proceeds to step S53, where an inquiry is made regarding whether the intended recipient answers the phone call (perhaps by way of a local PBX, such as PBX 9 in the office facility 7 of Figure 4). If the intended recipient answers the phone call, the process proceeds to step S55 where the phone call is connected to the intended recipient, and subsequently the process proceeds to step S65 where all the call processing is completed. Alternatively, steps S51, S53 and S55 may be performed by dialing directly the office number of the intended recipient, thereby bypassing the virtual call network processor 20. If the intended recipient is unavailable to answer, the local central office invokes a call forwarding operation that forwards the call directly to the Virtual Network Call Processor.

A majority of the time a given caller reaches a Virtual Voice Network is because the caller was calling a company rather then an individual. Although no two Virtual Voice Networks need to be identical, the majority of them greet the caller and instruct the caller to enter an extension number or choice. Some extensions do not process calls, some only process calls to a single number all of the time and some process calls to multiple numbers. CALL PULLBACK is invoked when a caller is offered other options after placing an unsuccessful call to a telephone not residing at the same location and not directly connected to the T3i Virtual Voice Network.

Call announcing and dialing multiple phone numbers are enhancements that the customer may or may not wish to use.

If the response to the inquiry in step S53 is negative, the process proceeds to step S57 where the call is forwarded (transferred) to the Virtual Network Call Processor 20, where the Virtual Network Call Processor 20 attempts to contact the intended recipient at one of the predetermined numbers stored at the Virtual Network Call Processor 20. In step S57 the caller is placed on soft hold, while an attempt is made to contact the intended recipient by way of an external line. The process then flows to step S59, where an inquiry is made regarding whether the intended recipient answers the call from the Virtual Network Call Processor at a first number stored in the Virtual Network Call Processor 20. Step S59 may have to be repeated if the intended recipient is not located at the first number and additional numbers are included in the intended recipient's profile that may be automatically checked by the Virtual Network Call Processor 20.

If the response to the inquiry in step S59 is affirmative, the Virtual Network Call Processor announces the call to the intended recipient in step S61. By announcing the call, the intended recipient has the option to receive the telephone call from the calling party, or have the Virtual Network Call Processor inform the calling party that the intended recipient is unable to receive the call. By announcing the call to the intended recipient, the intended recipient knows how to answer the phone, for example, when the calling party's call is taken off soft hold, and connected to the intended recipient's telephone. After step S61, the process proceeds to step S65, where call processing is completed and subsequently the process ends. On the other hand, if the response to the inquiry in step S59 is negative, the Virtual Network Call Processor invokes the CALL PULLBACK mechanism where the call remains on soft hold, and an external line (either the same external line as before, or another line) is used to attempt to contact the intended recipient at the next number identified in the profile of the intended recipient. Using this

example, the CALL PULLBACK mechanism would remove the caller from soft hold and announce that the called party was unavailable and offer further options, one of which could be to dial another number. This process of tracking-down the intended recipient proceeds until all of the candidate locations have been exhausted, at which time the Virtual Network Call Processor either takes a voice mail message, asks the calling party if they would like to identify another person to whom to route the call, etc. Of course, if the CALL PULLBACK mechanism successfully contacts the intended recipient, and the intended recipient decides to receive the call, the caller is then taken off hold, and connected to the intended recipient. Subsequently, the process is completed in step S65 and the process ends.

Application delays are timing values set in the Call Processor portion of the Node. Delays described in the Appendix are used to detect critical tone cadences that the Central Office provides to the Node equipment. These cadences indicate specific call conditions such as a ringback tone indicates that a called number is ringing, and a busy indicates that the called party is busy.

Cadence values are normally set by selecting a PBX type and making modifications to the equipment as needed. As it is not known what PBX type a given Central Office would have or what effect the state of repair or software level would have on the cadences provided, a starting point is to choose the PBX closest to the one which was part of a first node implemented by a user of the system.

After adjusting the Call Processor's cadence recognition to the first Central Office PBX, new adjustments can be made as Central Offices are added and tested to make sure that the new adjustments work with previous Central Office PBXs'. Fail-safe mechanisms are included as features of the CALL PULLBACK to catch any caller that hit an unexpected cadence. These fail-safe mechanisms allow callers the options of reaching a live operator or leaving a message as well as providing first hand intelligence regarding what happened and where an unexpected event

occurred. As the system matures, the fail-safes are not needed as frequently because the system's attributes will become more completely characterized with time.

Problems chiefly occur in areas where the Call Processor detects an answered condition while monitoring a single interrupted-ringback and with slow-busy and fast-busy cadences. When a call is screened, the equipment looks for acceptable cadences for a single interrupted ringback, slow-busy, fast-busy or that the call has been answered. To process a transfer application-delay, indexes are referenced that show the maximum and minimum ON/OFF periods for any tone. If the tone cadence detected does not comply with the ranges set for single ring back, slow busy or fast busy, the Call Processor determines that the call has been answered and the call transfer is completed.

In the case of dead air, such as the Central Office dropping the call, the caller would be removed from soft hold and a fail-safe mechanism would take over. One problem that occurs is when a Central Office recording is played such as an all circuits are busy or that the person being called is out of the area or unavailable. The Call Processor detects an answer and sound, such as someone speaking, and completes the transfer. These occurrences can be kept to a minimum and caller frustration reduced by the following methods:

Prior testing and identification - this allows the network designer to inform the customer of a potential problem and make needed changes or record a special greeting so that the caller has the opportunity to record a message or go to an operator before the called party's number is dialed;

Pulling the call back before the far end recording is played; and

Most customers and callers are used to the recordings being played and are not troubled by them. If the tone cadences are within acceptable ranges, call screening by the called party may be employed and the call accepted or rejected.

Figure 7A is a flowchart of a process for identifying local telephone equipment attributes,

such as frequency and cadence. The process begins in step S661 where the called party is dialed by a node. Subsequently, the process proceeds to step S663, where the cadence and frequency information from signals produced by the local telephone equipment is observed by the node. When observing the cadence and frequency information, the cadence and frequency information is characterized for subsequent processing. The process then proceeds to step S665, where application delays are identified that correspond with the frequency and cadence information that was characterized in step S663. The process then proceeds to step S667, where the node takes appropriate action for the call based on predefined custom parameters and/or reacts to the cadence events, where the reaction is a function of detecting which signals are in fact produced by the local telephone equipment. Subsequently, the process ends.

Figure 7B is a flow chart of a process used by a NODE-OVERTURE Call Processor to screen calls. The process begins in S71, where the call is transferred and the NODE-OVERTURE Call Processor dials the called number and begins looking at tone patterns received from the local telephone equipment. Subsequently, the process proceeds to step S72, where an inquiry is made regarding whether the tones that are received comply with the ranges set by delays 49, 50, 51 or 52, as identified in the appendix attached hereto. If the response to the inquiry in step S72 is negative, the process proceeds to step S73, where the call is considered to have been answered, and subsequently the process ends. However, if the response to the inquiry in step S72 is affirmative, the process proceeds to a ring back inquiry in step S74, where an inquiry is made regarding whether the received tones comply with the ranges set by delays 53, 54, 55 or 56. If the response to the inquiry in step S74 is affirmative, the phone rings and a ring back is monitored. The ring back minimum and maximum tone off period are included in the appendix. The process then concludes.

However, if the response to the inquiry in step S74 is negative, the process proceeds to step S76, where a slow busy inquiry is made. The inquiry in step S76 inquires whether the tones

comply with the ranges set by delays 69, 70, 71 or 72. If the response to the inquiry in step S76 is affirmative, the process proceeds to step S77, where the call is pulled back and the NODE OVERTURE Call Processor speaks the name and condition or greeting and subsequently the process ends. However, if the response to the inquiry in step S76 is negative, the process proceeds to the fast busy inquiry in step S78, and an inquiry is made regarding whether the tones comply with the ranges set by delays 73, 74, 75 or 76. If the response to the inquiry in step S78 is affirmative, the process proceeds to step S79, where the call is pulled back, and an indication is spoken indicating that the call is "invalid" and then the process ends. However, if the response to the inquiry in step S78 is negative, the call is answered in step S80 and then the process ends.

Figure 7C is an exemplary display of information that would be displayed on a monitor screen for a situation where a ring-no-answer operation fails. Application delays that produce a failure are first tested by assigning a mailbox that dials the problem telephone number to a special area code specific object. The port specific print tone trace is activated and the node's Call Processor is called through the port in question. The port will have a speakerphone butt set placed on it so that there is an audible awareness on the part of the person performing the task so as to determine what events are in fact occurring. When answered, the test mailbox is dialed and the tone on and tone off events are monitored in milliseconds as shown on a computer screen. In Figure 7C, a display 701, which, may either be a simultaneous display, or printout or otherwise of a stream of code, illustrates tone information that is displayed for a ring-no-answer operation that fails. As shown in annotation 702, the caller enters DTMF digits, which are the digits associated with the phone to be contacted. In annotation 703, the NODE OVERTURE monitors the line for a tone associated with a dial tone and then subsequently detects the presence of the dial tone. Once the dial tone is detected, and annotation 704 shows that the NODE OVERTURE dials the number associated with the DTMF digits. Annotation 705 indicates that the NODE OVERTURE ignores the first change in tone for a predetermined period of time. Subsequently,

as indicated in annotation 706, the NODE OVERTURE monitors call progress tones from the PBX so as to determine the status of the called extension. Finally, in annotation 707 a failure is indicated when the NODE OVERTURE detects an answer condition because one of the tones of the PBX does not conform to the delays in the application delay table, listed in the appendix.

The tone values in the print tone trace of Figure 7C may be modified to the correct values by using the appropriate commands. For example, in the case of Figure 7C, the application-delay indexes that refer to the error received are indexes 50 and 54, included in the appendix. Note that the failure occurred when the PBX sent a TONE ON for 790 ms. The NODE OVERTURE was set to expect a TONE ON (ring back) for no less than 800 ms and no greater than 1200 ms. This range between 800 ms and 1200 ms is referred to as the "window". In Figure 7C, the window for the silence period (TONE OFF) between adjacent rings is set to no less than 2800 ms and no greater than 3400 ms. The TONE OFF values are within that window.

Figure 8 is a block diagram of an intelligent network of Virtual Network Call Processors (20A, 20B, 20C, 20D) connected together, as shown, in a ring configuration, although other configurations may be performed as well, such as a star or non-geometric specific interconnected configuration. Each of the Virtual Network Call Processors 20A through 20D is configured as several interconnected nodes, a node being a PBX having a Call Processor. The Virtual Network Call Processors 20A-20D are the respective hubs, and thus each serves a different geographical area. As a consequence, a source communication device 1A may connect to the Virtual Network Call Processor 20A (or other Virtual Network Call Processor 20B-20D) by way of the local telephone equipment 30A, or directly to the Virtual Network Call Processor 20A. The Virtual Network Call Processor 20A may then route the information (voice, or other type of data, such as image data, facsimile data, etc.) via the other Virtual Network Call Processors (20B-20D) and then to a destination facility 7 either directly from Virtual Network Call Processor 20D, or via the local telephone equipment 30B.

Because the respective hubs 20A-20D are digitally linked, via dedicated point to point connections or by the use of VPNs' (Virtual Private Networks) or through a gateway server over the Internet, no charges for long distance services are required, although the network of Virtual Network Call Processors 700 certainly could charge a fee for such services or other fee-based links may be used as well.

An example application of the network architecture of Figure 8, might be if either an e-mail message, a facsimile message or other message such as a digitized voice, video or data file were intended to be left with a person in Florida (serviced at Virtual Network Call Processor 20A), a copy of that message may be routed through the network of hubs 20A-20D and to the destination facility 7. Since the connections are by way of dedicated links or VPNs, there are no long distance charges. In addition to the data relaying service, each of the respective Virtual Network Call Processors 20A-20D may also provide the standard call processing features described in Figures 4-5, for example.

The inventive system may include a CALL PULLBACK mechanism that employs a primary rate interface (PRI) that is compatible with National ISDN standards deployment of Simplified Message Desk Interface (SMDI).

The mechanisms and processes set forth in the present description may be implemented using one or more general purpose microprocessors programmed according to the teachings of the present specification, as will be appreciated to those skilled in the relevant art(s). Appropriate software coding can readily be prepared by skilled programmers based on the teachings of the present disclosure, as will be apparent to those skilled in the relevant art(s). The present invention thus also includes a computer-based product which may be hosted on a storage medium and include instructions that can be used to program a computer to perform a process in accordance with the present invention. The storage medium can include, but is not limited to, any type of disk including floppy disk, optical disk, CDROM, magneto-optical disk, ROMs,

RAMs, EPROMs, EEPROMs, flash memory, magnetic or optical cards, or any type of media suitable for storing electronic instructions, either locally or remotely.

Obviously, numerous modifications and variations of the present invention are possible in light of the above teachings. It is therefore to be understood that within the scope of the appended claims, the invention may be practiced otherwise than as specifically described herein.

CLAIMS:

1. A virtual network call processing system, comprising:

a communication line interface configured to be connected to a source terminal and receive a calling message from the source terminal directed to an intended recipient;

a call processor with a call pullback mechanism including,

a data processor, and

a computer readable memory having computer readable instructions encoded therein that when executed by said data processor implement a local telephone equipment characterization mechanism that characterizes signaling attributes of signals produced by local telephone equipment that service different geographical locations at which the intended recipient may be located; and

a signal determination mechanism configured to determine whether the signals provided by the local telephone equipment have at least one of a frequency and cadence associated with a signal event that includes at least one of a fast busy signal, slow busy signal, ringing signal, answered signal, and ring-no-answer signal.

2. The system according to Claim 1, further comprising:

an error handling mechanism configured to process the calling message when the signal determination mechanism fails to determine that the signal event occurred.

3. The system according to Claim 1, wherein:

said signal determination mechanism includes a software tool programmed to recognize the at least one of the frequency and cadence associated with the signal event from signals associated with the local telephone equipment.

4. The system of Claim 1, wherein:

said computer readable memory includes

an intended recipient profile, having a first destination number and a second destination number, and

said call pullback mechanism further includes a recipient contact mechanism being configured to attempt to first contact said intended recipient via an external line at the first destination number, and if not present, being configured to attempt to contact said intended recipient at the second destination number.

5. The system of Claim 4, wherein:

said signal determination mechanism includes a tone detector being configured to detect when said calling message on said communication line is answered, or said signal event occurs; and

said system further includes a call announcing mechanism configured to execute a whisper transfer to the called party that allows the called party to accept or reject the calling message.

6. The system of Claim 5, wherein:

said tone detector being configured to notify the recipient contact mechanism when said communication line is not answered so said recipient contact mechanism proceeds to contact said intended recipient at said second destination number.

7. The system of Claim 1, wherein:

said local telephone equipment characterization mechanism includes a frequency characterization mechanism, configured to characterize a frequency of the signals produced by

respective of said local telephone equipment.

8. The system of Claim 1, wherein:

said local telephone equipment characterization mechanism includes a cadence analyzer configured to analyze a cadence of respective of the signals provided by respective of said local telephone equipment.

9. The system of Claim 8, wherein:

said local telephone equipment characterization mechanism includes a frequency analyzer configured to analyze a frequency of the signals provided by respective of said local telephone equipment.

10. The system of Claim 1, further comprising:

a signal feature normalization mechanism, including an application delay adjustment mechanism configured to adjust respective application delays in said call processor so as to standardize signal attributes provided by respective of the local telephone equipment.

11. The system of Claim 10, further comprising:

another data processor and another computer readable memory configured to implement another local telephone equipment characterization mechanism and another signal feature normalization mechanism, said processor and computer readable medium being connected to said another processor and said another computer readable medium by an intercity communication link.

12. A method for processing a call in a virtual network call processing system,

comprising the steps of:

receiving a calling message from a source terminal directed to an intended recipient;

retrieving a data profile of the intended recipient from a computer readable medium;

identifying a number to contact the intended recipient via a local telephone equipment;

characterizing signal attributes of signals provided by the local telephone equipment;

initiating the call on an external line with said number at said local telephone equipment;

normalizing the signal from the local telephone equipment;

transferring the calling message if the call is accepted by the intended recipient, but retaining the calling message for future processing if the call is not accepted by the intended recipient.

13. The method of Claim 12, further comprising the steps of:

identifying another number in said data profile of said intended recipient;

calling on at least one of the external line and another external line, said another number at another local telephone equipment;

normalizing a signal from the another local telephone equipment;

transferring the calling message if accepted by the intended recipient at the another number, but retaining the calling message for further processing if not accepted.

14. The method of Claim 13, further comprising the step of:

producing a tone on said external line and detecting when said external line is answered.

15. The method of Claim 12, wherein:

said characterizing step comprises characterizing a frequency of the signal provided by the local telephone equipment.

16. The method of Claim 12, wherein:

said characterizing step comprises characterizing a cadence of the signal provided by the local telephone equipment.

17. The method of Claim 16, wherein:

said characterizing step further comprises characterizing a frequency of the signal provided by the local telephone equipment.

18. The method of Claim 12, wherein:

said normalizing step comprises adjusting respective application delays so as to standardize signal attributes of the signal from the local telephone equipment.

19. The method of Claim 12, wherein:

said calling step comprises passing the calling message from a first hub to a second hub, prior to reaching the local telephone equipment.

20. A computer readable medium encoded with computer readable instructions for use in a system having a communication line interface configured to be connected to a source terminal and configured to receive a calling message from the source terminal directed to an intended recipient, said computer readable instructions when executed by a data processor implement a system comprising:

a local telephone equipment characterization mechanism that characterizes signaling attributes of signals produced by local telephone equipment that service different geographical locations at which the intended recipient is located;

a signal determination mechanism configured to determine whether the signals provided

by the local telephone equipment have at least one of a frequency and cadence associated with a signal event that includes at least one of a fast busy signal, slow busy signal, ringing signal, answered signal, and ring-no-answer signal; and

a call pullback mechanism configured to call an intended recipient and transfer a calling message to said intended recipient if said signal determination mechanism determines said call is answered, but not transferring said calling message if said signal determination mechanism determines that said call is not answered.

21. A virtual network call processing system, comprising:

means for receiving a calling message from a source terminal directed to an intended recipient;

means for identifying a number to contact the intended recipient via a local telephone equipment;

means for characterizing signal attributes of signals provided by the local telephone equipment;

means for initiating a call on an external line with said number at said local telephone equipment;

means for normalizing the signal from the local telephone equipment; and

means for transferring the calling message if the call is accepted by the intended recipient, but retaining the calling message for future processing if the call is not accepted by the intended recipient.

CALL PROCESSING SYSTEM, METHOD AND COMPUTER PROGRAM PRODUCT

ABSTRACT OF THE DISCLOSURE

A system, method and computer program product implement a Virtual Network Call Processor with a CALL PULLBACK mechanism for providing a type of screened call transfer. Callers, while attempting to contact an intended recipient, have their calls sent to the Virtual Network Call Processor, which places the caller on soft hold while attempting to locate the intended recipient. The Call Processor uses another external line to call the intended recipient at one of a number of predetermined locations identified by stored numbers where each number is serviced by perhaps different local telephone equipment having different characteristics and attributes. The CALL PULLBACK mechanism is used to identify signaling attributes of signals provided by the respective local telephone equipment, by analyzing frequency and cadence information from the signals and normalize the signals so as to detect a status of the Call Processor's attempt to reach the intended recipient. The signaling attributes and customer-specific information are controlled by objects, which are well thought out preprogrammed and proven software constructs that simplify programming and ensure reliable operations. The calling party is kept on soft hold while the intended recipient of the call is attempted to be contacted at the different locations. If the CALL PULLBACK mechanism determines that the signals provided by the local telephone equipment, after being normalized, indicate the intended recipient does not pick up the call, the CALL PULLBACK mechanism attempts to reach the intended recipient at another one of the numbers, all the while the calling party is kept on soft hold. In this way, the global Virtual Network Call Processor, is capable of servicing not only individuals and companies serviced by a single PBX with a call process, but also for any number of other users not serviced by the PBX.

APPENDIX

APPLICATION DELAY TABLE.INDEX DELAY (msec.)

0	0
1	7000
2	5000
3	500
4	5000
5	1200
6	1000
7	30000
8	800
9	4000
10	6000
11	1000
12	1000
13	10000
14	20000
15	25000
16	2000
17	20
18	1000
19	200
20	240
21	100
22	140
23	200
24	260
25	300
26	400
27	260
28	460
29	800
30	900
31	1200
32	1500

33	2000	
34	2100	
35	2700	
36	3900	
37	4000	
38	6000	
39	1000	
40	4000	
41	10000	
42	1500	
43	1500	
44	2000	
45	500	
46	1700	
47	700	
48	500	
49	2500	Maximum tone on period for any tone. When ringback, busy or fast busy is encountered this delay is used to determine whether the tone on is a valid tone. When an encountered tone on is longer then this delay it is assumed to be an answer.
50	50	
51	4600	Maximum tone off period for any tone. When ringback, busy or fast busy is encountered this delay is used to determine whether the tone off is of valid duration. When tone off is greater then this value it is assumed to be an answer.
52	40	
53	2500	Maximum tone on period. This is the longest tone on event that will be considered as ringback. This application delay is used to determine whether the tone on cadence event being monitored is ringback. If the tone on event is longer then this application delay it is assumed to not be ringback.

54	800	
55	4600	Ringback maximum tone off period. This is the longest tone off event that will be considered as ringback. If the tone off event being monitored is shorter then this application delay it is assumed to not be ringback.
56	2000	Ringback minimum tone off period. This is the shortest tone off event to qualify as ringback. This application delay is used to determine whether the tone cadence being monitored is ringback. If the tone cadence being monitored is less in duration then this value it is assumed to not be ringback
57	2300	
58	20	
59	15000	
60	20	
61	500	
62	300	
63	500	
64	300	
65	3200	
66	2800	
67	280	
68	120	
69	600	
70	400	
71	600	
72	400	
73	320	
74	180	
75	320	
76	180	

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